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Focus on Acoustics

Acoustical Solutions Round-Up 2022
Improving Acoustics in Home, Office, and Workspaces
By J. Martins

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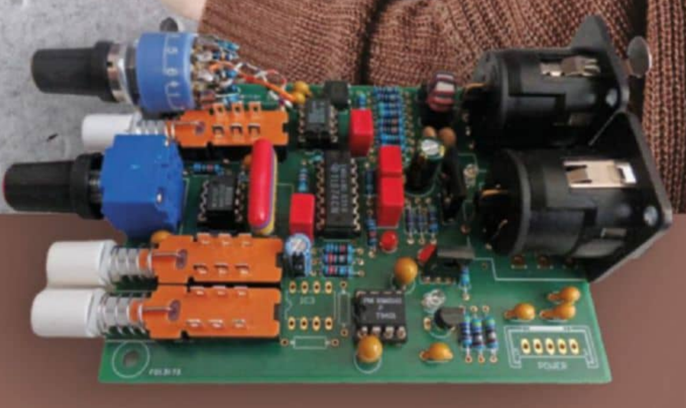
Acoustic Diffusion Research
By Norman Varney

You Can DIY!

Distributed Subwoofers for Problem Rooms
By Thomas Perazella

Sound Control

Reverberation: Control and Measurement
By Richard Honeycutt



You Can DIY!

"Pass-Able" Fully Discrete Line Stage
By Aren van Waarde

Building a Guitar-Controlled Synthesizer: VCA & VCF
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You Can DIY!

DC-Coupled Microphone Preamplifier
By Michel Nieuwenhuizen

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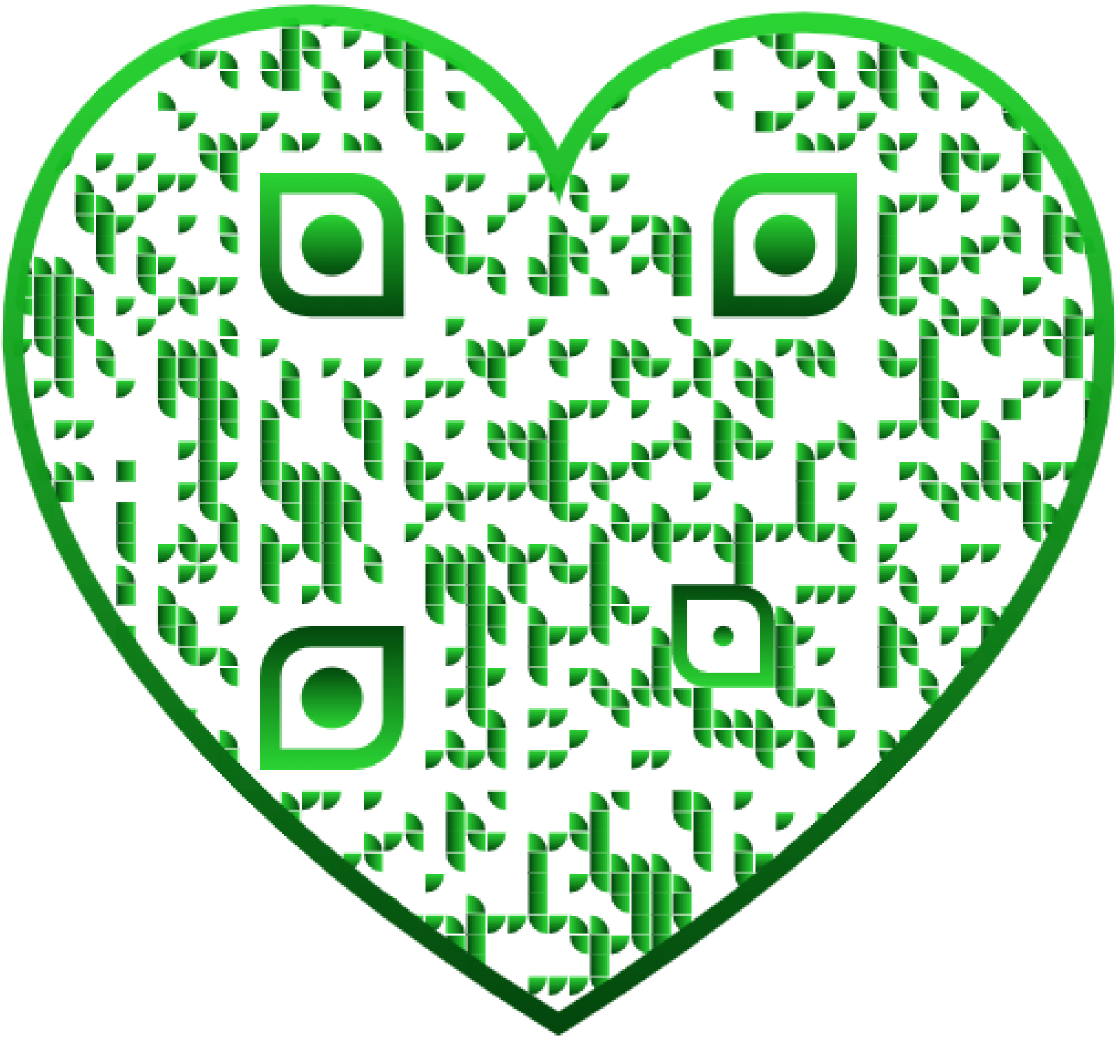

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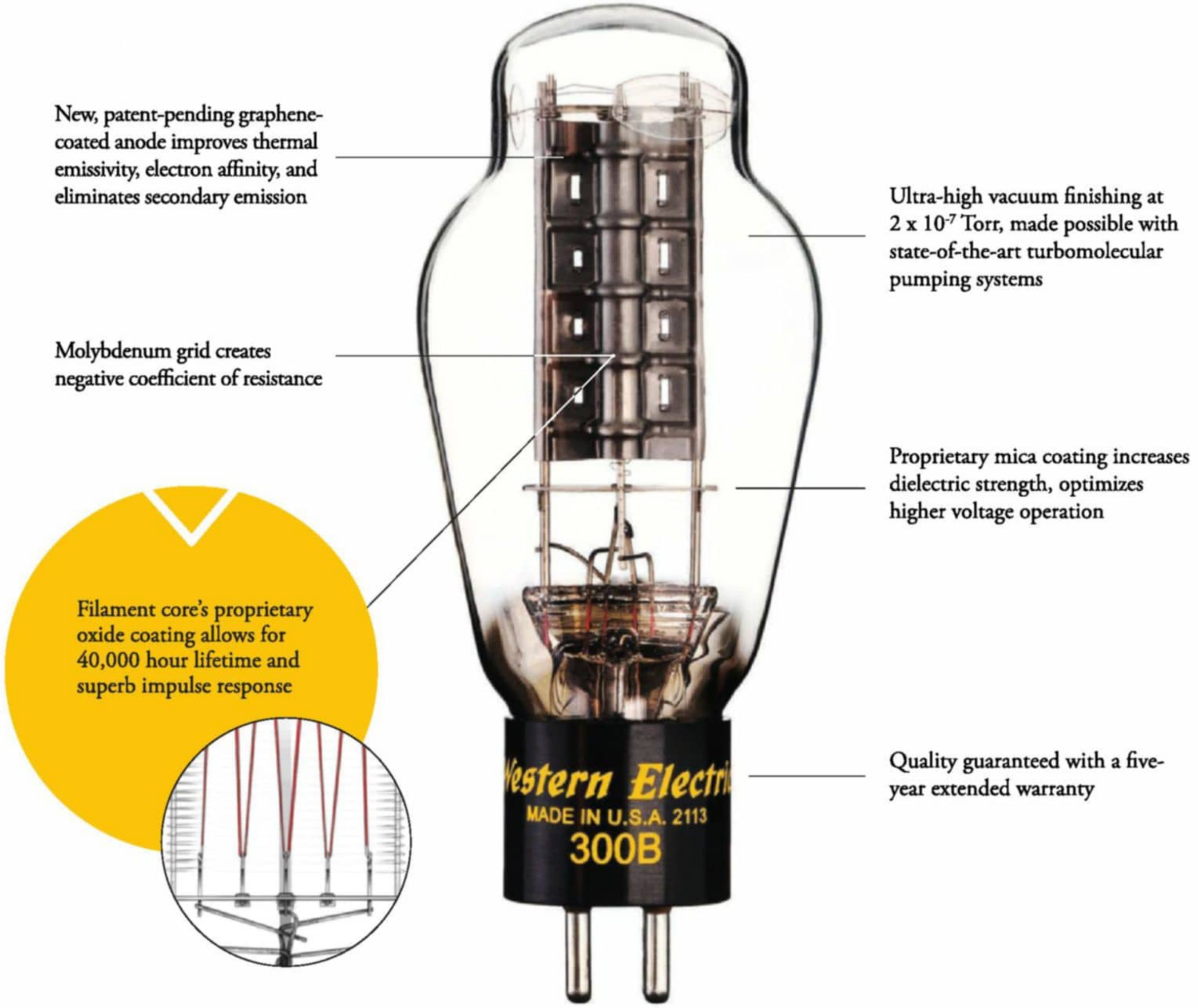

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The Secrets of Acoustics Revealed

Every time I have to stop writing and wrap up another issue of *audioXpress*, one of the common feelings—I guess, also common to most magazine editors—is the mortification about all the things that have been left out. And it is a feeling that repeatedly haunts me every year with our Acoustics Focus issue.

In this month's Acoustics Market Update, there's a few examples of products and solutions from companies I would really have liked to detail a bit more, such as the wonderful work of Offecct, a Scandinavian design company that continuously offers architects the possibility to create unique rooms by combining sound absorbers in different design materials, helping to reduce noise and create balanced rooms and spaces. And the work of Techlite and its remarkable customized structures using open-cell melamine foam, which we have selected to illustrate, but didn't have the option to explain.

And I would love to have written more about the remarkable work of Atanis Zhelev and Maria Korolova, the husband/wife team behind SHION, a new range of active solutions that explore the convergence of active noise canceling, immersive sound technologies, sensors, artificial intelligence-enhanced DSP, and engineered materials, to go beyond conventional acoustic treatment systems.

I have been writing about how technology is currently reaching the turning point, when these things are possible and commercially viable. And SHION is attempting to go even further by creating a "Sound Bubble in Motion" approach, using AI to analyze a listening room, locate and motion-track the listener. Fascinating stuff. But I guess we will have to return to the topic in a future article focused exclusively to active acoustics.

Even with all the power of these available technologies, we still depend on the fundamental understanding of sound propagation and acoustics. And I believe more articles about acoustical science and advancements in acoustics should be featured in a publication such as *audioXpress*.

However, I am very pleased when we have valuable contributed pieces from industry friends that reflect their work and continuous efforts and research to offer solutions that can benefit everyone. One such piece is offered in this issue, written by Norman Varney, someone who tirelessly combines his work on listening room design with the development of practical solutions for common problems and never stops sharing what he learns.

And of course, there's Richard Honeycutt's regular Sound Control column for *audioXpress*, where he "shamelessly" shares the secrets of acoustics that apparently "everyone" knows but very few discuss.

Sometimes I feel like giving up on reaching out to acoustics experts to invite them to share their knowledge and their work with our valuable audience. Because I know that—when they write—they will either submit the deepest scientific treaty about mathematical modeling of noise propagation, or chapter one of "the basics of sound." And probably the Acoustical Society of America and the Audio Engineering Society are to blame, because they focus on deep-level academic scientific papers and forget about the praxis and the essential link to audio product development and the industry—normally for fear of being "commercially" tainted—which is pure nonsense.

This magazine could be just the right platform to promote practical work—be it designing anechoic chambers, modeling and simulation of phase modulation in parametric loudspeaker arrays, or particle models for accurate binaural rendering of spatial audio (hope the authors are reading this...)—and create a much-needed direct contribution to the evolution of audio engineering and audio product development, particularly now that we are embracing practical applications with active acoustics, noise cancellation and acoustics modeling, and designing sound reinforcement systems for immersive audio reproduction of object-based audio content.

I truly believe there's much to be done in this domain to better connect acoustics with the overall audio industry. And the invitation remains for those who understand how much the audio industry could benefit from a deeper connection with acoustics.

J. Martins
Editor-in-Chief

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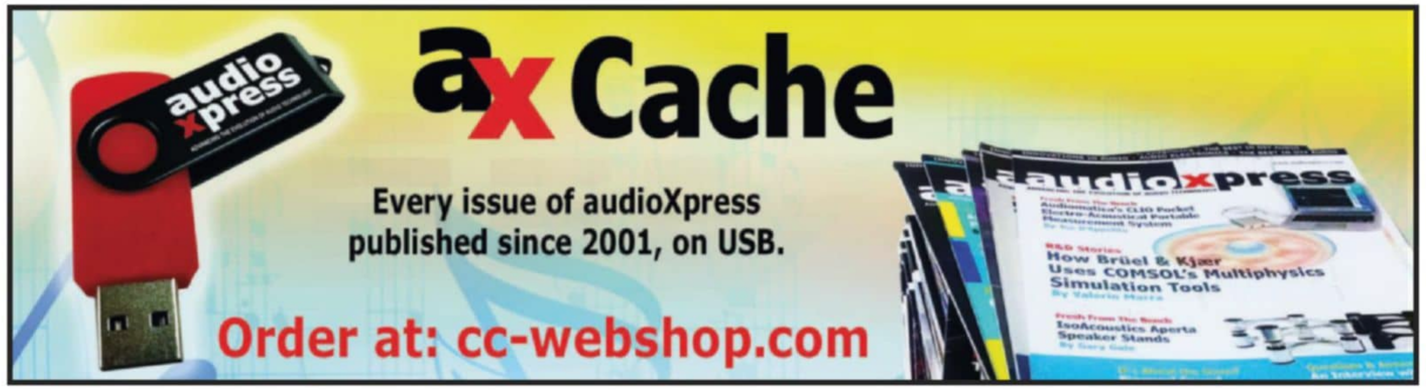

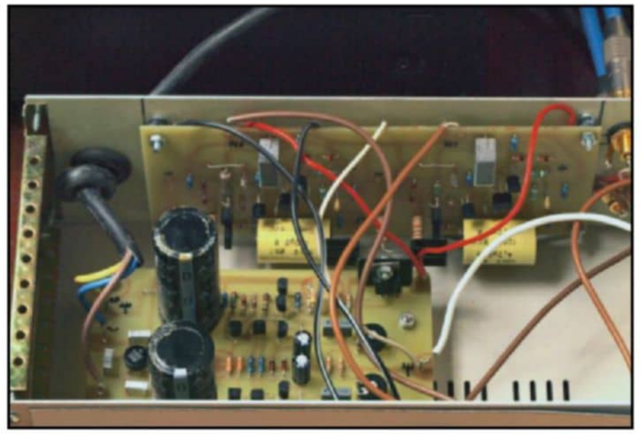

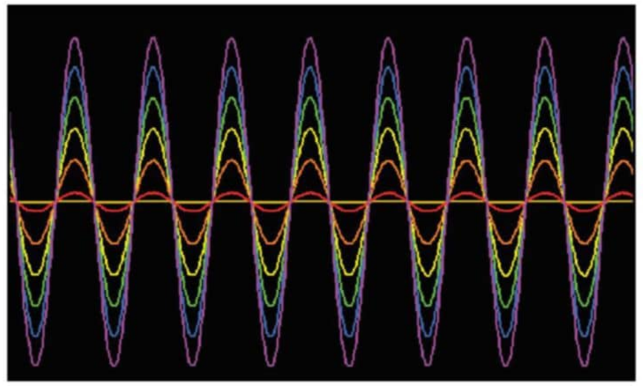

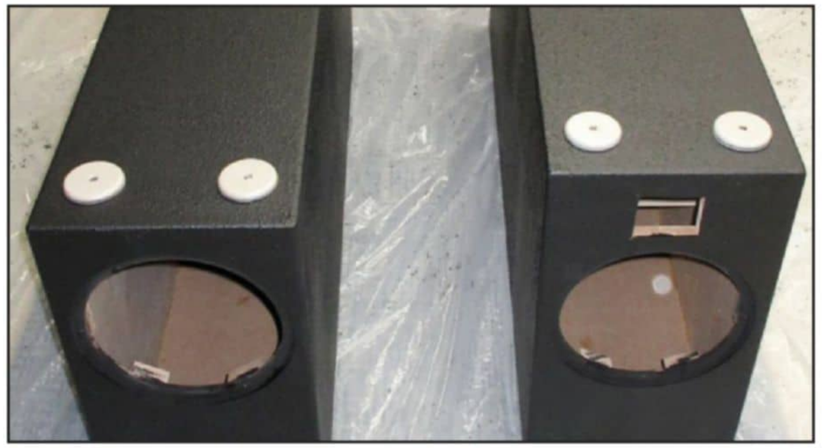

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By Michel Nieuwenhuizen

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By Aren van Waarde

To play music via a commercial power amplifier, the author built a line-level preamplifier with an "old-fashioned" gain stage using only discrete transistors, inspired by a Nelson Pass article. Combined with a quiet power supply and a small PCB-type power transformer, the results surprised positively.

60 Building a Guitar-Controlled Synthesizer
VCA & VCF

By Ethan Winer

Ethan Winer learned how to design electronic circuits by piecing together smaller building blocks using knowledge that he gained from various electronics magazines and from the application notes from semiconductor makers. For the past four issues our author has shared the modules he used to build a guitar-controlled synthesizer. In this final article in the series, he describes how to make two types of voltage controlled amplifiers plus the classic Moog low-pass filter.

COVER ART

Output Eco
Acoustic Panels
Credit: De-Fi Acoustic

Focus on Acoustics

MARKET UPDATE

8 Acoustical Solutions Round-Up
Improving Acoustics in Home, Office, and Work Spaces

By J. Martins

As *audioXpress* does every year, we have compiled another Acoustical Solutions Round-Up highlighting companies and products that define market trends in acoustics.

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By Norman Varney

This article shares Norman Varney's journey on developing new diffuser solutions, and lessons learned while designing countless critical listening rooms around the world. He was actively involved in multiple acoustics standards committees, including for testing procedures, and he explains why diffusers continue to be misunderstood and many times wrongly used.

36 Distributed Subwoofers for In-Room Low-Bass Problems

By Thomas Perazella

After discovering room modes in his listening room, Thomas Perazella opted to solve his problem by building and placing multiple subwoofers at different locations in the room to create a greater distribution of modes.

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SOUND CONTROL

32 Reverberation: Control and Measurement

By Richard Honeycutt

Since the construction of the first performance venues, acousticians have struggled with acoustical interference, reverberation, and echoes. In this article, our acoustics expert discusses ways to control sound and how to accurately measure it.

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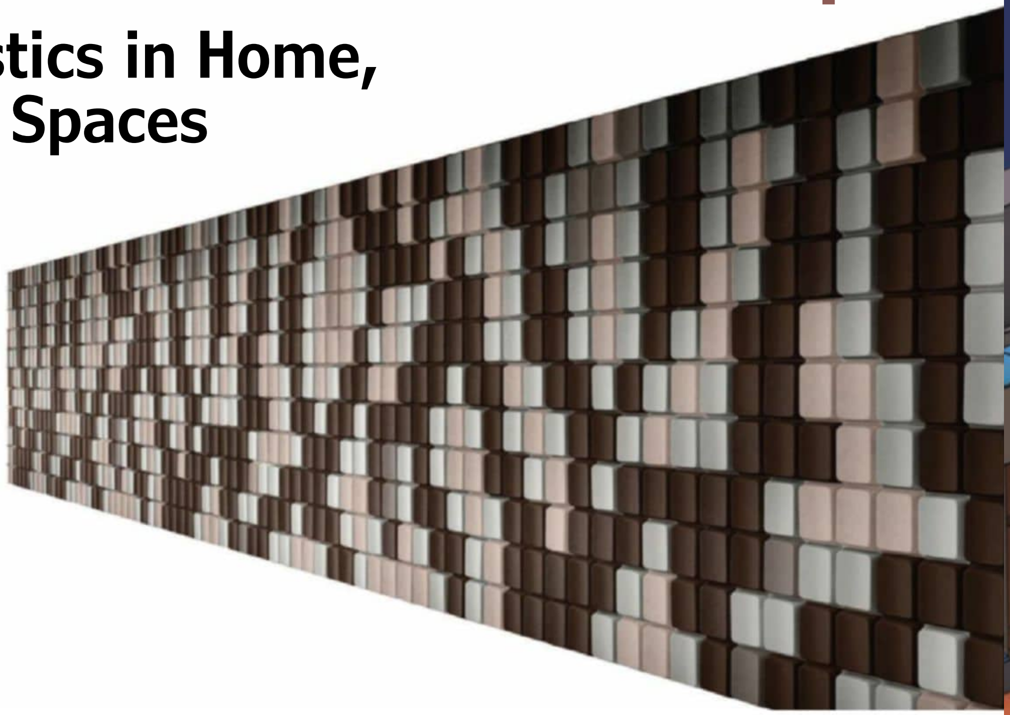
Acoustical Solutions Round-Up

Improving Acoustics in Home, Office, and Work Spaces

As *audioXpress* does every year, we have compiled another Acoustical Solutions Round-Up highlighting companies and products that define market trends in acoustics.

By
J. Martins
(Editor-in-Chief)



Offecct is a Scandinavian furniture design company that develops acoustic solutions for inspiring meeting places in the home, office, and public space. Soundwave is a well-known collection of modern acoustic wall panels that apart from its acoustic properties also brings character to the interior. This sound absorbing panel Soundwave Pix is designed by the French designer Jean-Marie Massaud.

Each year, there are increasing numbers of acoustical products on the market helping to tackle the most demanding projects. Particularly for studio environments, performance and rehearsal spaces, and cinemas (commercial and residential), the level of attention to detail and the science applied to such installations continues to allow improving results. For the past two years, a lot of the leading companies in acoustical solutions for those applications have also taken the time to expand their research and design new ranges of products, but in fact many of them simply reacted to the sudden need and new requirements that were motivated by the global pandemic—with millions of people—including musicians, composers, music producers, audio engineers, and any audio-related professions—having to work from home.

This situation significantly accelerated the trends that had been gradually building to improve acoustics in home studios, with versatile and practical solutions that don't necessarily require redoing a building's foundation or building floating cages inside one's home. At the same time, since schools had closed and students required remote learning, and everyone needed to communicate with colleagues and work remotely, people noticed how their home environments were challenging due to interfering and undesired noises, and how the acoustics were not ideal for communications, remote presentations, and recordings—something that had never been in people's consciousness or even a concern.

For that reason, in our 2020 Acoustic Solutions Round-Up, we focused on those home environments, listening spaces, and adapting spaces for communication and remote collaboration applications. In 2021, as the world asymmetrically and gradually evolved to new ways of dealing with the virus and some regions of the world reopened, everyone realized we needed to also readapt the spaces where we would normally remain for most hours of the day, when outside our homes. And in our 2021 Acoustic Solutions Round-Up we looked at new priorities in acoustic materials that made offices or hotels improved spaces to work and communicate, while also introducing the need for cleaning and disinfection of the acoustical solutions used in those spaces.

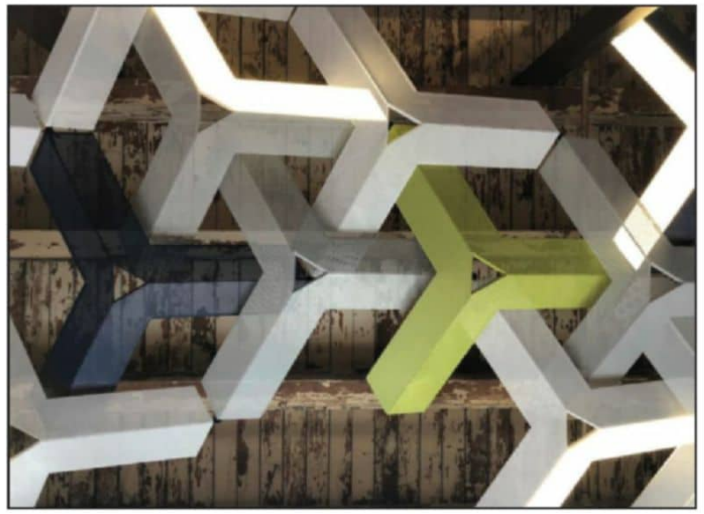
Over the past two years, many companies around the world have used that opportunity to propose new solutions that feature antimicrobial treatments and easy to disinfect materials, while also highlighting key sustainability concerns and adopt eco-friendly recycled and recyclable materials. All trends that remain and will continue to evolve as the people in charge of designing buildings, rooms, and spaces in general are looking at acoustics differently.

Another core trend that will also remain top of mind for all of us, is the need to be in "healthier" public spaces in general, and that includes the offices to which we will eventually have to return. This means—as I noted last year—not only having improved acoustics in those spaces for new forms of communication and collaboration, but also understanding the way acoustical treatment solutions are designed and installed, taking into account their impact in the overall indoor air quality.

Sound Conditioning

I have now started to attend trade shows again and I have discussed these trends with companies involved in acoustics and acoustical solutions and materials, as well as audio manufacturers in general, and everyone agrees that things have changed even in the relatively conservative market of high-end listening spaces, with those considerations in mind. The change was already under way to address home studios and listening rooms in general with more aesthetically pleasing installations and materials, away from the foams, cardboard, or traditionally "uncleanable" custom wood and fabric panels and bass traps. Now, the owners of those spaces, including professional studios that need to deliver the best possible results, are looking for "healthier" acoustic treatment and insulation options.

And talking about insulation and sound and noise control products, now that people think it's safe to have contractors back into their homes again—and now that they don't need to be there all the time—there is exploding demand for such projects since people realized what they have to deal with when neighbors or even their family is at home and they really need to work.



Techlite melamine foam acoustic panels shown here are used to create an architectural surface, balanced and suitably-tuned.

By far, the largest market for insulation treatment materials was commercial buildings, followed by industrial applications. That was a market that was already seeing strong growth in result of ever more stringent government regulations toward reducing noise pollution and enhancing green building construction. Those investments are now increasingly considered part of the need to address recognized adverse health effects. But now, there will also be the need to readapt a totally new class of spaces and to seriously address the home market. If until now the residential market was not so significant for insulation products, the evolving requirements will contribute to making those applications one of the most important ones, at least in volume terms.

For manufacturers of acoustic insulation and noise control products, including the large industrial and chemical giants, this will also involve the need to adapt their business in order to address completely new challenges and evolving safety regulations.

For this 2022 Acoustical Solutions Round-Up, we have also selected a few interesting products indicate that the market is responding to all those trends. We also highlight companies and solutions representing the hybrid class of "decorative" products that are not really acoustic treatment products per se, but that when applied to large surfaces, walls, and ceilings, or applied in great numbers as space dividers, do become acoustically effective for the core purposes of living and working in those spaces.

In this article, we also explore the growing segment of sound isolation booths, space dividers and other approaches for workspaces that also tick the boxes of sustainability and creating attractive aesthetic options.

And finally, we have selected a few new examples of active acoustics, a new approach that is becoming increasingly approachable for demanding applications with the need for variable acoustics. A topic that *audioXpress* intends to expand on very soon in a dedicated report.

Acoustiblok Corp., a NASA spin-off company, manufactures and markets innovative products including the Acoustiblok viscoelastic polymer-based soundproofing material and Acoustiblok Wall-Cover, made and sourced in the United States.

SHION

Adaptable Acoustic Treatment with Active Acoustic Panels

Recently presented at AXPONA 2022, SHION active acoustic panels are designed by a new company from Vienna, Austria, that is exploring advanced patent-pending technologies both in hardware and software, using a combination of smart sensors and transducers. SHION means "Sound of Stillness" and the name was derived from the Japanese term SHIN-kan (the silence of natural stillness) and NO-oto (sound).

SHION is developing a modular system of smart acoustic treatment panels for music listening spaces, where each panel features a motion-sensing algorithm that uses AI and ultrasound to cancel noise, calibrate the acoustic performance of any space, and deliver the optimized sound to the listener's position in real time—while offering an elegant visual statement that can complement any interior design.

Multiple frequency actuators produce fine-tuned vibration cycles over the surface of the panels, a specially engineered carbon-wood composite—sending a targeted response through the actuated surface that mediates and enhances the sound. The vibration cycles of the acoustic surface control the amount of sound absorption at different frequency ranges from 200Hz to 5000Hz.

With lessons learned from smart speakers and modern noise-cancelling algorithms, SHION's approach focuses on tracking the listener's position, and optimize the sound around the space, creating a "sound bubble in motion." Wherever the listener is detected becomes the sweet spot, enhancing the performance of any audio system playing in the room. According to the company, the system also cancels unwanted noise and is able to change the acoustic response of the space by loading impulse responses with one click in the app.

Physically, the SHION system is designed to look as a standard acoustic panel, with easy-mounting wall panels or free-standing modules that can replace or complement passive acoustic treatment in any room. A minimum of three panels are required and the ability to change the acoustic response depends on both the room acoustics and number of panels. All panels come with the complete computing system, embedded electronics and all necessary wiring.

After the initial setup, the system self-calibrates in real time, and is able to generate up to 11dB of active noise cancellation. This makes it suitable for both home theaters/listening rooms as well as larger rooms looking to cancel noise and create focused acoustic zones.

Atanis Zhelev and Maria Korolova—the husband/wife team behind SHION—have roots in high-end design. Both are part of the world-renowned Zaha Hadid architectural studio, which has extensive experience developing innovative interior solutions in challenging spaces. In 2020, Korolova and Zhelev established SHION to address conventional acoustic treatment systems. "No acoustic system before could cover both issues: acoustics and aesthetics," says Zhelev. "The biggest challenge usually is in big spaces with reflective surfaces, panoramic glazing, home theaters, minimalistic design approach and lack of performance of conventional systems."

Seeing an opportunity in the fields of active noise canceling and immersive sound technologies, they brought together engineers from the fields of IoT, automotive, and aeronautical design. SHION's debut product, previewed at AXPONA 2022, was highly praised.

The proprietary material developed for SHION's modules is called Active Grid Monocoque (AGM)—a reference to the micro-structure of the engineered material. "It offers the sensual and aesthetic quality of natural wood, and can be 3D-shaped in an organic manner. Every panel is acoustically optimized and treated with fire retardant coating," says Korolova. "Any surface can be activated, but for excellent sound quality and frequency range you need a carefully arranged ratio of hard and soft composites to properly distribute the sound and work with different frequencies."

The SHION solution is distributed in North America by VANA, Ltd., based in Nesconset, NY.

www.shion.at

SHION's active acoustic panels are fully integrated systems that include sensors, actuators, amplifiers, and embedded processing.

BAUX

Climate-Neutral Acoustic Design

Baux is an interesting company from Stockholm, Sweden, that addresses interior design, architecture, and acoustics, always following a climate-neutral philosophy in its building materials and products. It even promotes a useful *Handbook of Acoustic Design* intended for architects and interior designers, which can be downloaded for free.

BAUX designs, produces, and markets construction materials that meet the contemporary expectations of architects, engineers, and builders—without compromising safety and environmental standards. It offers acoustic treatment products, designed primarily with public, commercial spaces in mind.

For those acoustic panels, BAUX perfected a series of environmentally friendly, recyclable materials, starting with a combination of wood and wool, cement and water that provides great absorption properties but is also moisture-regulating, heat-accumulating and fireproof. Another material is the 100% bio-based BAUX Acoustic Pulp, combining the performance properties of sound absorption, safety, and durability with sustainability and modern aesthetics. BAUX Acoustic Pulp is recyclable, biodegradable, 0% production waste, fire retardant, and water repellent.

For is latest range of acoustical solutions, BAUX created the Acoustic Felt panels using a next-generation Oeko Class I textile and GRS-certified recycled PET bottles sourced from recycling entities across Europe, accredited by IOAS (https://ioas.org). The Acoustic Felt panels are 100% recyclable and available in a palette of six colors, including three greyscale tones and the more saturated hues of coral and jade. The textile base is also upcycled and 100% traceable in partnership with The Loop Factory.

www.baux.com

BAUX acoustic Wood Wool panels are used to create large walls and focus rooms in hotels.

Acoustic Felt panels use recycled PET here applied as space dividers for offices.

Artnovion | Kuietly

Creating Safe and Healthy Workspaces

Kuietly is a new brand created with a mission to drive innovation and engineer acoustic performance, creating unlimited design possibilities and the best solutions for office spaces. In partnership with Artnovion, the renowned specialist in acoustic design and technology, Kuietly combines the best of design, acoustic performance, and quality finishes, in an easy-to-use, sustainable range of state-of-the-art products, designed to efficiently target common acoustic issues found in office environments.

The Kuietly Keen line of dividers available in Desk and Floor panels.

Kuietly sources recycled and sustainable materials for all its products, in an effort to minimize environmental footprint. And all its designs combine acoustic technology with contemporary aesthetics in products that use recycled PET-based materials, upcycling potential ocean waste.

The recently launched brand is for now focusing on a core line of products, with a modular range of space dividers for offices, SweetSpot isolation booths and a range of suspended cloud panels to tame office noises. Complete projects are also designed using a selected range of Artnovion acoustic panels and acoustic wallpapers, all of which have the use of sustainable materials and recycled PET in common.

The company's range of dividers includes the Kasual, Keen, and Klever lines, the first using fully recycled PET with a solid Birch plywood structure, while the second uses the same core materials, with lacquered, wood veneer, and laminated finishes with antibacterial, antifungal, and antimycotic protection. It is equipped with a recycled PET absorption core and manufactured from Baltic Birch Plywood sourced from sustainably managed forests. The Klever dividers are available in woven fabric or faux leather, also with antibacterial, antifungal, and antimycotic protection.

These dividers are designed to create a sense of visual and acoustic privacy in a shared open-plan office while adding to the overall acoustic quality. Available in a large array of sizes to fit any workstation, the dividers simply clamp to the edge of a table, or are available in free-standing desk and floor screens—some with acrylic guards. Basically, all panels are Class C and offer absorption in a range of frequencies from 250Hz to 8000Hz with an NRC of 0.65.

The Kuietly SweetSpot

More than a booth, the SweetSpot acoustic PODs provide a new option for co-working environments and open offices, allowing for added privacy, a special focus session or important meeting. The booths again feature recycled PET and durable Birch plywood carefully sourced from sustainably managed forests. The door detailing is available in a choice of black and white vinyl finishes.

Due to their multiple layers, these pods are high-performance soundproofed workstations, ideal for work environments in which the high noise level drastically interferes with productivity. With the door closed, the SweetSpot booths offer up to 30dB of sound insulation and excellent interior acoustics for conference calls and voice recording. Each pod also features optimized ventilation provided by two ultra-quiet fans, energy-efficient LED lighting, a built-in power socket, and USB charging, with a height-adjustable rest or seat.

www.kuietly.com

Kuietly Sweetspot isolation booths for open office environments

Artnovion

New Cinema Series Range for Media Rooms

Artnovion envisioned a new way of creating home entertainment spaces and launched a new Cinema Series range of acoustical solutions for a new and optimized experience. This new range brings new acoustic tools to designers and integrators that simplify the interior and acoustic design of any space, totally separating acoustic performance from the aesthetics of the space.

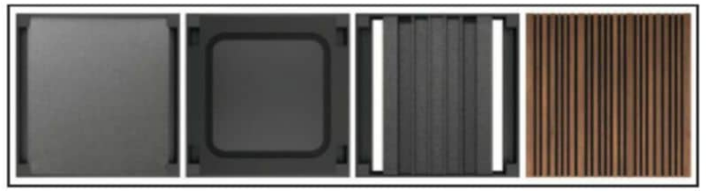
"Over the past few years, there has been a distinct market tendency towards residential acoustics. For reasons we all know, people have been spending more time at home, delving into new entertainment systems, or simply spending more time listening and expanding their room setups," says Jorge Castro, Artnovion founder and CEO.

These products form a modular panel system that is provided with four different types of core: an Absorber, a Diffuser, a Bass Trap, and a Speaker core. All the cores can be fitted with different cover designs that can be freely removed, swapped, and replaced between cores. All panels are equipped with a discreet modular installation system.

These modular acoustic panels also allow enthusiasts to create the perfect space for a stereo system and easily reconfigure and expand the acoustic treatment for a 5.1 surround, by changing the location of the panels in the room. Modifying the cover designs over the cores completely changes the space and expands the acoustic treatment along with the room system. If the interior design needs to be overhauled, the covers can be simply swapped to create a whole new ambiance.

This new range comes in a selection of distinct wooden and fabric designs, including Valley CS, a highly efficient fabric absorber that plays with geometric patterns to create abstract shapes, Helen CS using a neutral fabric with impressive absorption, Komodo CS wooden absorbers with an acoustically optimized perforation pattern, and Siena CS hybrid wooden acoustic panels that enable the creation of custom spaces, combining high performance and minimal aesthetics.

www.artnovion.com



Artnovion Cinema Series panels are available with absorber, bass trap, diffuser, and speaker cores, as well as a range of covers.

Vicoustic

Broadband VMT Mega Bass Traps

In 2022, Vicoustic celebrates its 15th anniversary of developing acoustic innovations, during which the Portuguese company has become a global brand, pioneering sustainable acoustics, producing over one million acoustic panels, and developing thousands of projects worldwide.

The most recent development from Vicoustic includes two next-generation broadband bass traps, designed to deal with low end, medium, and high frequencies in a single product, creating a valuable solution for home and project studios, as well as listening spaces or media rooms.

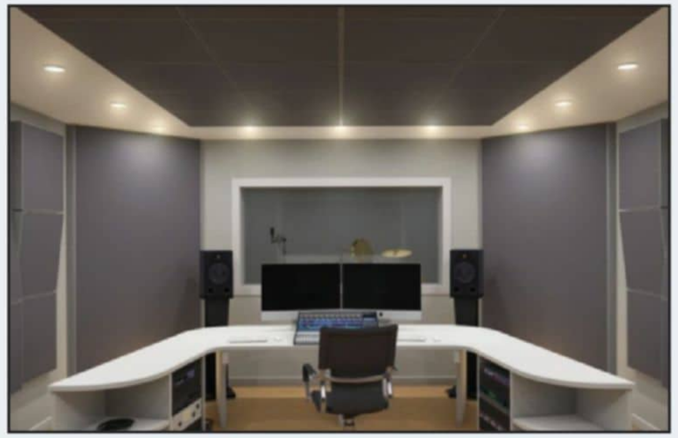
Mega Bass Trap VMT (Virtual Material Technology) helps deal with all frequencies from 80Hz onward. Designed for corners, this specific version can also be installed in the ceiling edges, using the Mega Bass Trap VMT Horizontal Kit optional mounting hardware.

Mega Bass Trap VMT XXL is a broadband solution that goes even further by helping to deal with the full frequency range from 50Hz onwards. Its extended performance and ability to control most of the audible frequency range makes it a valuable solution to handle the acoustic defects of most rooms, such as flutter echoes and speaker boundary interference.

Vicoustic was able to develop these products once again using recycled PET materials, and the bass traps are certified for Indoor Advantage Gold Building Materials, by SCS Global Services, meeting indoor air quality recommendations.

Available in five selected VMT colors and three different frame finishes, one box of Mega Bass Trap VMT or XXL includes two units, with the VMT front and the VicPET Wool filling to place in the interior to maximize the absorption. Like any other VMT product, the front is totally customizable, enabling printing, or color of preference.

www.vicoustic.com



Vicoustic Mega Bass Trap VMT are broadband bass traps.

Vicoustic

Expanded VicBooth Ultra Sound Insulation Range

Updating its standard catalog of first-approach acoustic panel solutions sold globally, Vicoustic launched the VicBooth acoustic cabin and sound insulation booth concept in 2016. The DIY modular design was so popular that Vicoustic quickly launched the VicBooth Ultra modular system with more options for multiple applications. Recently, Vicoustic expanded the range with two larger sizes—3m×3m and 3m×4m—and introduced the new VicBooth Ultra Shop online store.

The Vicoustic VicBooth Ultra modular acoustic booth has quickly become a very popular option for many companies, studios, and professionals that use it in their recordings and other daily applications. From voiceovers recorded from home, to large companies working in speech recognition and audiobook projects, or studios in need of additional separate recording options, the range proved extremely effective and shipped to all corners of the globe.

Renowned for its acoustic solutions, Vicoustic was able to create an effective sound insulation booth at an unbeatable price. Any of the modular VicBooth Ultra models can achieve noise reduction levels over 30dB with only 1.1" thick walls, measured according to ASTM standards. The booths' modules use acoustic surfaces combining melamine, MDF, and are coated with VicPET Wool, an eco-friendly non-woven textile with great acoustic performance, predominantly made from recycled plastic bottles.

Additional updates to the ceiling beams reinforced the booth structure, enabling larger sizes up to 3×3 and 3×4 options. Larger sizes and extra room interiors pave the way for a wider range of applications, particularly for music production studios and teamwork.

To enable a better understanding of all the VicBooth Ultra options and range, Vicoustic also decided to introduce a dedicated online store, together with an online configurator available for customized options.

The Vicoustic VicBooth range is certified by SCS Global Services for Indoor Air Quality Performance Standards, meaning it meets the Furniture Emissions Standard recommended for the open plan and private office workstation parameters. With these certifications, Vicoustic products now contribute to credits in green building rating systems.

Vicoustic products are distributed in the United States by Focal Naim America.
www.focalnaimamerica.com
www.vicoustic.com

Vicoustic expanded the VicBooth Ultra range with bigger 3m×3m and 3m×4m sizes.

Flex Acoustics

Variable Acoustics for Music Venues

Flex Acoustics was founded in 2005 by Niels Werner Adelman-Larsen, based on the discovery of high sound absorption properties of inflated flexible membrane structures under certain conditions and the possibilities derived from these. After many years of research and development, a certifiable product became available in 2013.

When patents were confirmed, DTU Innovation, a venture fund company at the Technical University of Denmark under Seed Capital Denmark, and three other private entities invested in Flex Acoustics, paving the way for the development of the aQflex and aQtube acoustic treatment products.

Following a period exploring those two acoustic treatment products in many successful installations, Flex Acoustics has now announced evoke, a powerful variable acoustics technology, which transforms a hall acoustically in a matter of seconds.

Using the evoke technology, multipurpose halls can present a wide variety of styles of music with literally optimal acoustics: from choir and chamber music to brass bands to rock concerts and even cinema. All with ideal reverberation time in every frequency band to ensure an optimal sound experience and performance.

The system consists of numerous modules that are embedded in large, or entire ceiling and wall surfaces. Each module, measuring 240cm × 60cm, is a basic building block that can be angled to ensure a more sound-diffused room. Each module also features its own dedicated low-voltage motor, which allows the front to open for absorption or close to become a diffusor, creating a shorter or longer reverberation in the room.

These variable modules are connected in groups, and each group allows a different result according to a preset for each acoustics environment or basic style of music. There is no limit to the amount of groups that can be allocated in a system, and each module could be operated individually.

The surfaces for these evoke panels feature different perforation patterns and can use wood, fabrics, or any other materials, creating three-dimensional textures that acoustical consultants, architects, and designers can freely explore.

Walls and ceiling, or key parts of each, become fully absorptive or reflective, at the flick of a switch, depending on which panels should be open/closed in the various presets.

www.flexac.com

The evoke system from Flex Acoustics allows variable acoustics and can come with an optional service package, which ensures a long-term lifespan.

WhisperRoom

30 Years of Modular Sound Isolation Booths

Since 1990, WhisperRoom has manufactured portable and affordable sound isolation enclosures, becoming a leading name in the category, going beyond standard vocal booths. A simple and original concept derived from the desire to play a saxophone in an apartment without disturbing neighbors, resulted in the first WhisperRoom, and the first portable and modular sound isolation booth on the market. It quickly became the foremost sound isolation enclosure being distributed to thousands of clients around the world.

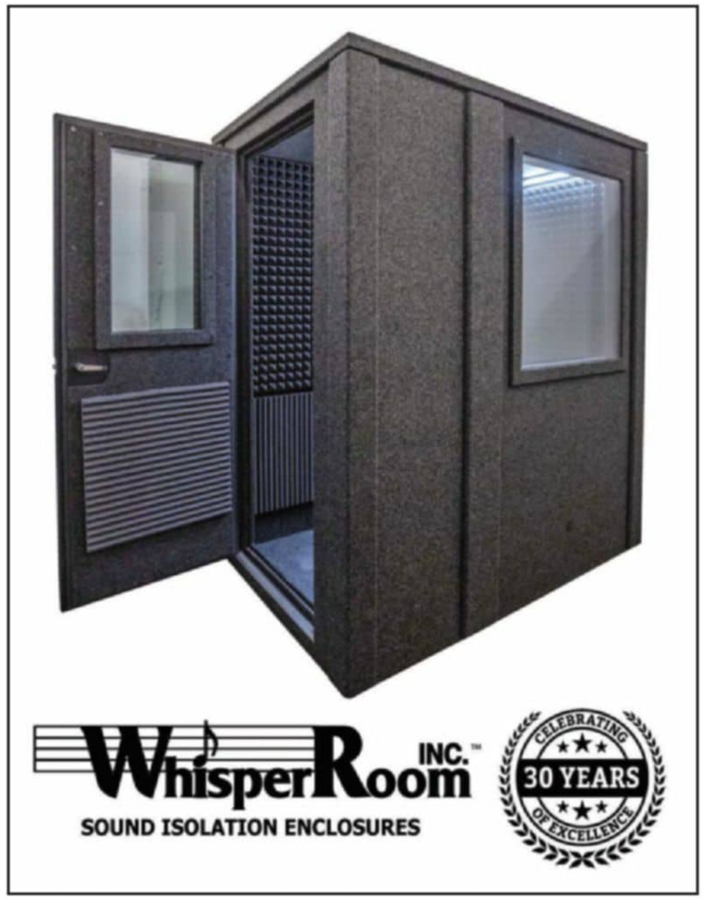
From its humble beginnings more than 30 years ago, WhisperRoom, Inc. has grown into the world's premier supplier of modular sound isolation booths. Now, the company offers 26 different sound booth model sizes that are available in two isolation levels—Standard (single-wall) and Enhanced (double-wall). Additionally, there are more than 16 optional features that can be added to customize the sound booth for its application, including interior acoustic treatment with products from Audimute and Auralex Acoustics. This flexibility easily meets commercial and residential for recording, music instrument practice, translation, voice-over work, research, and many other applications.

WhisperRoom also offers audiometric booths that are designed for audiologists and speech-language pathologists. These products have met OSHA standards and ANSI 3.1 requirements of "acceptable ambient noise levels for audiometric testing rooms" as defined by the Acoustical Society of America (ASA).

All WhisperRoom components are kept in stock and are ready to ship via common carrier freight. Components are individually packaged, stacked on pallets, and wrapped with bright red plastic. Custom crates are also available for transport and storage.

The modular design of a WhisperRoom is intended for the end-users to self-assemble. Assembly, disassembly, and relocation can easily be performed by two people and all the tools are provided. Each component is designed to be of manageable weight and size. Depending on the model, the typical heaviest single piece will weigh between 79lbs. and 120lbs. Ceiling and floor components are attached to wall components with hinged brackets. Stainless steel bolts and pre-installed inserts are used to attach wall components and seam seals to one another. The complete assembly is performed from the interior with minimal exterior space required.

www.whisperroom.com



WhisperRoom's modular sound isolation booths allow many sizes and meet all application requirements, available in two isolation levels.

Output | De-Fi

Eco Acoustic Panels Modular System

Output, the Los Angeles, CA-based company that evolved from music creation software tools into studio furniture and stands, and surprised with its Frontier nearfield active studio monitors designed by Barefoot Sound, has also launched a modular acoustic panel system. The new Eco Acoustic Panels expand Output's studio hardware catalog and offer a complete high-performance, open acoustic treatment system, designed in collaboration with De-Fi Systems.

The Output Eco Acoustic Panels modular range—Absorbers, Bass Traps, and Diffusers—was built to create an optimal studio experience in standard rooms and home environments. In alignment with Output's ethos, the panels were intentionally designed with both top notch sound quality and aesthetics in mind. By making the mounting system part of the sonic design and utilizing air gaps, De-Fi reduced the amount of material required to get equivalent or better performance than much larger and chunkier solutions.

The choice of materials was another primary focus in the Eco Acoustic Panels. The full system was crafted with sustainability in mind, leveraging post-consumer recycled materials in each element. The hardwood walnut used for the connector plates are salvaged from urban forests in the United States, using wood that would normally be discarded. The Eco Panels are made primarily of post-consumer recycled plastic.

The patented system is designed to create an air gap between the Eco Panels and the surface behind, which in conjunction with the panel perforations, allow utilization of both sides, by absorbing both frontal energy and reflections. The mounting solution is designed for single and double stacking Eco Panels. This allows for customization of not only the amount of absorption desired, but gives the opportunity for unique visual color combinations.

A single stack Eco Panel system has an average NRC rating of 0.8, which is equivalent to most 2" thick flush mounted wall solutions. Stacked Eco Panels systems have an average NRC rating of 0.9, which is equivalent to most 3" thick flush mounted wall solutions.

The air gap and perforated PET felt panels used in both the Absorbers and Bass Traps remove unwanted reflections and low frequency build up at comparable RT60 values to much thicker, wall-mounted materials. The Diffusers, made from high-quality hardwood, are tuned to scatter mid-range frequencies for ideal sonic conditions.

Absorber panels are available in single- and double-wide columns, and in three colors (Granite, Charcoal, and Sunset). The patented Bass Trap uses Eco Panels for higher frequencies on the front surface, and handles frequencies behind down to 125Hz, featuring an acoustic membrane constructed from post-consumer recycled material, available in the same three colors. The quadratic diffusers are tuned to reflect mid-range frequencies between 847Hz and 3400Hz, and can be placed in horizontal or vertical orientation, available in Kodiak, Natural Walnut, and Driftwood Grey.

www.de-fi.com
www.output.com

Besides offering a flexible modular acoustic treatment solution, low environmental impact was a driving force behind the Eco Acoustic Panels.

Primacoustic

PrimaPro Room Treatment Calculator App

When it comes to acoustically treating a room, there is no one-size-fits-all formula. Factors such as room dimensions, available wall and ceiling space, and the intended application of the room can dictate the proper acoustic panel selection.

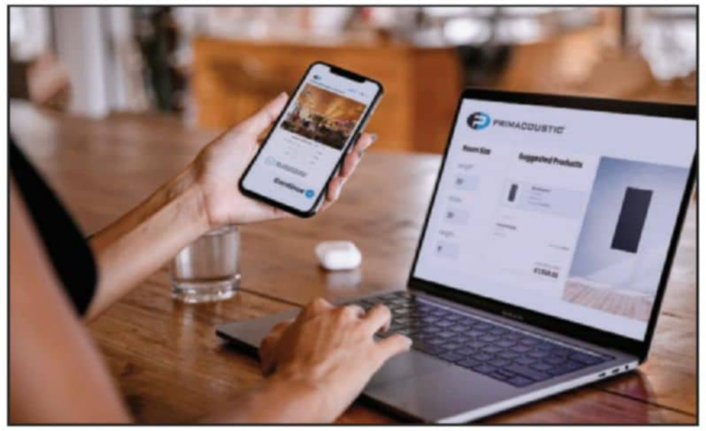
Primacoustic introduced the PrimaPro Calculator, a user-friendly, step-by-step tool that takes the guesswork out of an acoustic treatment project. The PrimaPro Calculator app was designed to suggest the best-suited acoustic treatment products based on application and dimensions, and will make it easy to explore the possibility of starting an acoustic treatment project, or expanding existing installations.

The PrimaPro Calculator app is designed to be easy to use, with the whole room analysis process only taking a few minutes. The user is first prompted to enter a few key details about their space: room type, dimensions, and panel placement (wall, ceiling, or both). The app does the background calculation and the user is presented with a complete package of acoustic treatment options.

Primacoustic is the acoustic treatment solutions company originally created in 2000 by Radial Engineering and quickly became an established brand in the professional recording market and home studios.

The app is available online here:
https://app.primacoustic.com/express-treatment

www.primacoustic.com



Odeon A/S

New ODEON 17 Software Released

Danish company Odeon A/S develops and distributes ODEON, one of the most popular and more powerful room acoustics simulation and measurement software. The ODEON software is used worldwide for prediction of room acoustics and PA-systems in concert and opera halls, theaters, worship spaces, sports stadiums, open-plan offices, restaurants, music studios, transportation terminals, industrial environments, and even outdoor areas with complicated geometry.

Results come as acoustic parameters, sound mapping, binaural or surround sound auralization allowing users to analyze results with their ears or present "soundscapes" of future rooms to customers.

Odeon A/S is located at DTU Science Park, which is part of the campus of the Technical University of Denmark, 10km north of Copenhagen. This is close to the DTU acoustic facilities, which include one of the world's best anechoic rooms with a free volume of 1000 cubic meters.

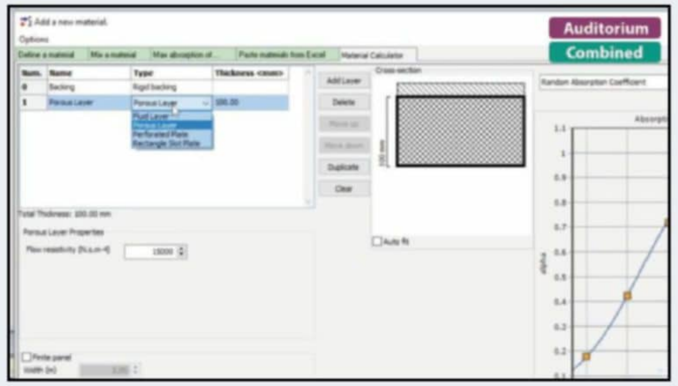
At the start of 2022, the Danish company released ODEON 17, a new version that includes attractive new features and speed enhancements. Available in Auditorium, Industrial, and Combined "editions," ODEON 17 now comes with a Material Calculator module, which makes it possible for acoustic, material companies and users to create their own materials. ODEON Industrial and Combined are equipped with a Sound Power Estimation tool, which allows estimating the sound power levels of sources, based on a number of SPL measurements. And the Auditorium and Combined editions include greatly improved tools for handling multi-source auralizations, making it a breeze to transfer a complete orchestra simulation from one of its many OdeonZipArchives.

On the interface improvements, source-receiver editing becomes even easier, most of the colors in layers, reflections and other illustrations can now be fully customized, loudspeaker cabinet illustrations can now be imported, and common directivity files are displayed in the 3DView.

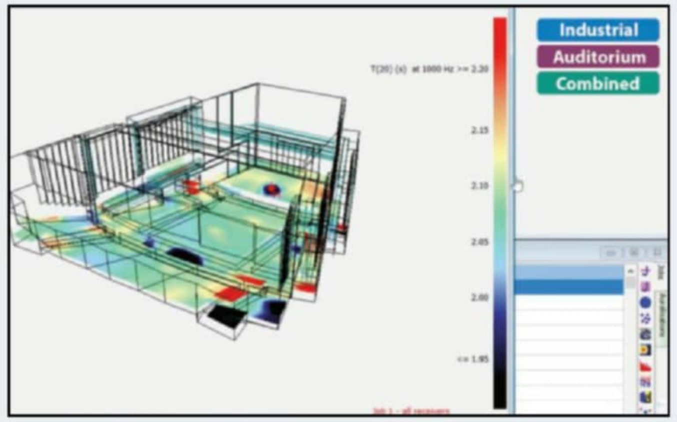
Odeon improved calculation algorithms for simulation of impulse responses of coupled spaces, and due to major changes in the way of handling rays, ODEON 17 outdoor simulations (e.g., a square surrounded by buildings in the city) have become easier than ever.

ODEON 17 also adds improvements to the measurement facilities. Measured impulse responses, room acoustic parameters and auralizations can be normalized to target conditions in terms of temperature (T) and relative humidity (RH%). As users can both measure and simulate the acoustics in ODEON, it has been made possible to easily compare the two. This can also be done with measurements made in another way or with another software. This is good for validating models when working with acoustical optimization projects in existing rooms. Those are available starting with the Basics edition

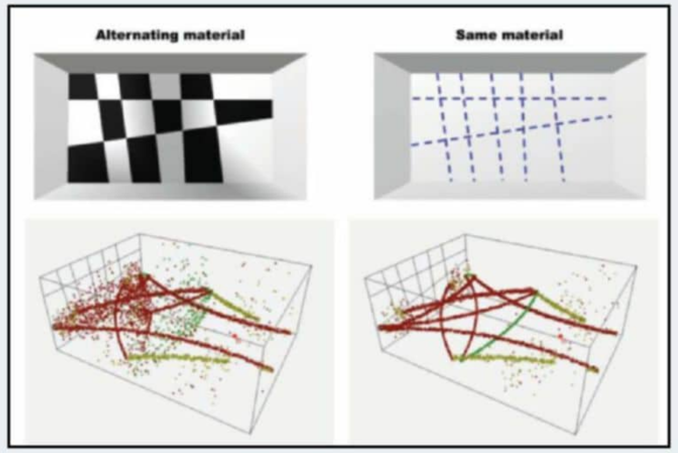
Odeon.dk



A Material calculator is included in ODEON 17 and allows estimating absorption of materials, such as porous absorbers (e.g., suspended ceilings), resonators (perforated or slit perforated), and thin membrane materials. You can vary the thickness and configuration by adding more than one layer. The tool will calculate the overall absorption coefficient of the compound structure.



The Auditorium edition of ODEON 17 is for a large set of room acoustical parameters based on the reverberation curve. It has a variety of graphical and auralization tools with an adjustable color scale.



ODEON will now calculate alternate material surfaces, which are in the same surface plane.

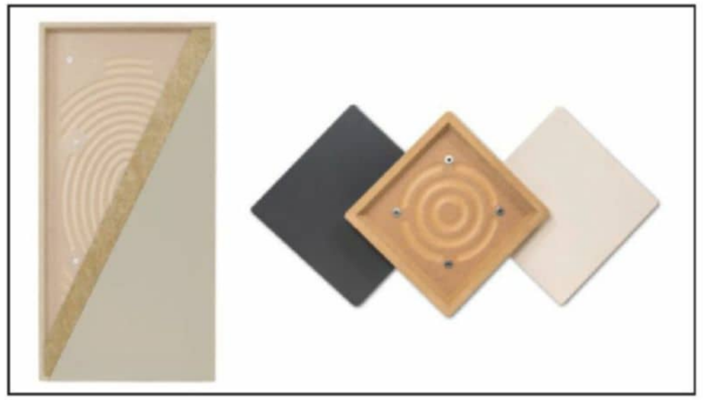
ProSoCoustic

WAVEPro Series with Interchangeable Fabric

ProSoCoustic's WAVEPro Series panels offer a unique blend of absorption and diffusion in one product. A diffusor pattern inside each panel allows greater absorption efficiency in the same amount of space. ProSoCoustic uses Rockwool Rockboard 40 stone wool, a mixture of basalt and steel slag, due to its superior absorption, health, safety, and sustainability factors over fiberglass or foam.

WAVEPro panels also include convenient machine screw inserts that provide simple installation via a single pair of 18" Z-Clips. Additional machine screw inserts provide easy access for additional Z-Clips, offering extra durability in high-traffic areas or when hanging panels from the ceiling.

Exclusive to ProSoCoustic's WAVEPro Series, the fabric is attached to the panel using a screen spline, which can be taken out for easy fabric replacement. Users can replace fabric if old fabric becomes damaged, a color change is desired, or if they want a custom-printed image printed on fabric.



The WAVEPro Series' frame includes diffusion designed right into the product. The frame is an UltraStock FR MDF panel that has a Class A Fire Rating, contains no added formaldehyde, and is FSC and Eco-Certified. Each panel is machine-cut to create a special design with the most effective diffusion properties.

ProSoCoustic offers the option to sew screen spline directly into custom-print fabric for an extremely simple fabric reinstallation. This enables users to safely store extra fabric in a weather-resistant, clean area and change their WAVEPro Panel's aesthetic as often as desired.

ProSoCoustic WAVEPro Series acoustic treatment panels are mold and moisture resistant, UL classified and ASTM E84 Class A fire-resistant, ISPM-15 certified for worldwide transport, and meet LEEDs requirements for environmental safety with FCS and ECC certification.

ProSoCoustic's WAVEPro Mini panels offer all of the same features as their larger counterparts but in a much lighter (and more affordable) package that can easily be installed in a diamond formation.

Whether upgrading a bedroom studio, home theater, vocal booth, streaming setup, or just trying to reduce sound buildup, WAVEPro Mini Kits are a perfect foundation to begin an acoustic treatment journey.

https://prosocoustic.com

Exclusive to ProSoCoustic's WAVEPro Series, fabric is attached to the panel using a screen spline, which can be taken out for easy fabric replacement.

Versare

Acoustical Partition Walls and Panels

From its homespun roots in Minnesota providing partitions and office cubicles to schools and churches, Versare has grown to be an international provider of solutions that create flexible spaces anywhere. Gradually, Versare evolved to offer an extensive catalog of solutions, including for conditioning acoustics in homes, classrooms, offices, warehouses, medical facilities, and more. Versare was also one of the first companies to sell these flexible space solutions online with modular, temporary, and semi-permanent walls, floors, rooms, and sound-absorbing ceiling panels.

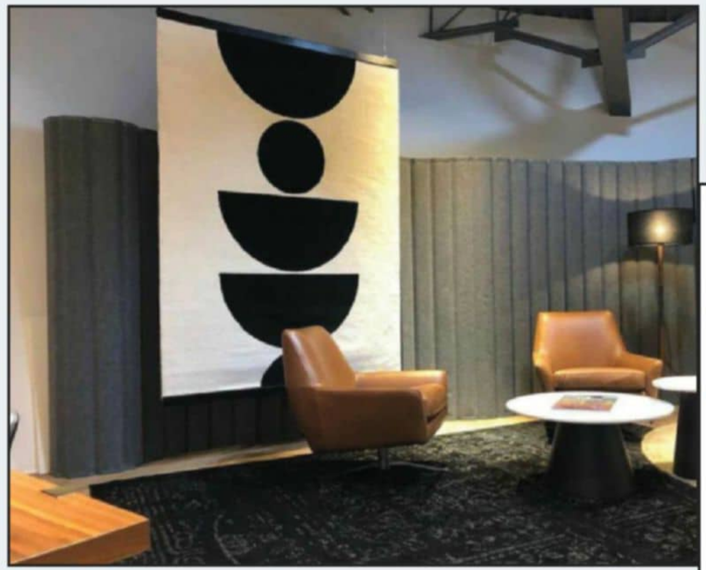
The company's latest portable SoundSorb VersaPanel Acoustic Partition was designed in response to the latest market trends and the needs to lessen the noise in workspace. The SoundSorb VersaPanel Acoustic Partition is constructed of high-density rigid polyester, reducing reverberation in any environment with NRC-rated material, up to 0.60. It is a simple solution to deploy, helps keep sound confined, and it folds flat for easy storage when not required.

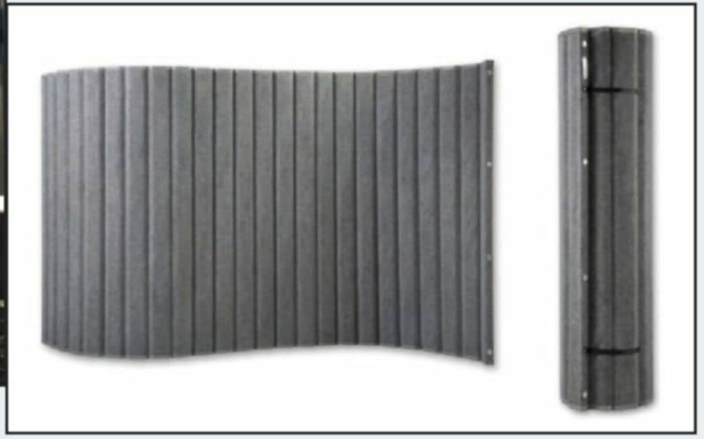
Versare SoundSorb VersaPanel Acoustic Partition

As Versare highlights, these partition walls are made with materials that won't fade or wash out and are eco-friendly, using at least 60% post-consumer recycled material, being solvent-free, and not using volatile organic compounds (VOCs).

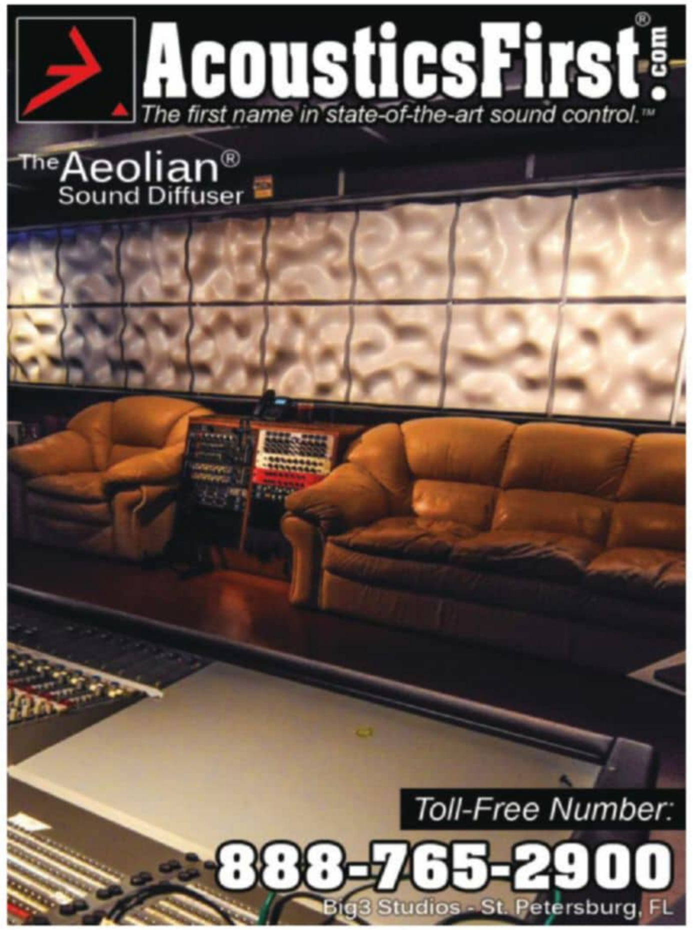
Another complementary solution in this same approach are the VersaPanel Acoustical Partition Walls, which Versare designed to reduce noise and separate spaces using acoustic-ribbed fabric and a heavy foam core that give this partition an NRC rating of 0.30. The partition meets the floor along its full width to help keep sound confined and can be arranged in an S, C, or L shape. Linking several units together helps in getting a sound-focusing backdrop that can also be easily rolled up for storage.

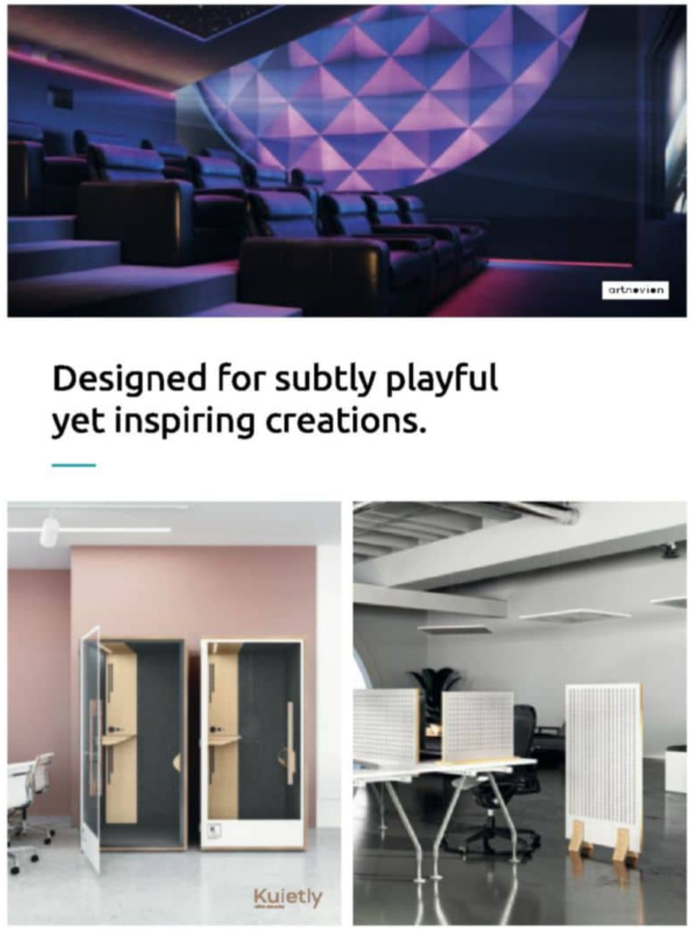
www.versare.com





VersaPanel Acoustical Partition Wall





pinta acoustic

New CONTOUR Ceiling Tile Patterns

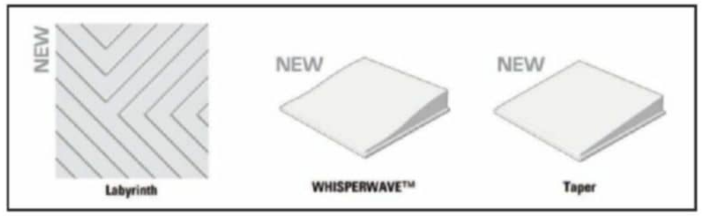
pinta acoustic's CONTOUR Ceiling Tiles are now available in three new patterns, including Labyrinth, Taper, and Whisperwave, providing a modern look and visual movement. According to the Minneapolis, MN-based company, the new ceiling tiles create modern effects while absorbing unwanted noise and are ideal for small- and large-scale indoor spaces. CONTOUR Ceiling Tiles are used in modern offices, lobbies, retail spaces, restaurants, entertainment facilities, worship centers, and more..

The new CONTOUR Ceiling Tiles are made with open-cell, mold-resistant WILLTEC foam, and are offered in two styles for easy, efficient ceiling grid or glue-up installation. CONTOUR tiles offer a noise reduction coefficient (NRC) up to 1.20 and CONTOUR tiles pass Class A per ASTM E 84, CAN ULCS-102 and University of Pittsburgh Toxicity of Smoke Emission tests.

Standard direct-apply, glue-up tiles are 24″ × 24″ (610mm × 610mm) in thicknesses of 1″ or 1-3/4″ (25mm or 44 mm). Suspended ceiling grid tiles fit in standard 15/16″ (24mm) grid systems and have a backerboard that blocks sound from traveling into adjacent rooms with a ceiling attenuation class (CAC) of 34. Ceiling grid tiles are 23″ (584mm) square with thicknesses of 1-3/8″ or 2-1/8″. CONTOUR Ceiling Tiles are available in natural white or light grey, and a water-based acoustic coating is available in standard, premium, and custom colors for easy cleaning and durability.

pinta acoustic is part of the pinta elements group serving the European and North American markets. Over the last 50 years, the company has gained extensive experience with acoustic and noise control products, thanks in great part to its SONEX series of classic acoustic products. pinta acoustic also conducts joint research with the Fraunhofer Institute to develop innovative solutions.

www.pinta-acoustic.com



The three new CONTOUR patterns provide a dynamic visual style and exceptional sound absorption.

Acoustics Vendor and Services Directory

Acoustiblok | www.acoustiblok.com
acouStaCorp | www.acoustacorp.com
Acoustic Fields | www.acousticfields.com
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Acoustics in a Box | www.acousticsinabox.com
Acoustic Manufacture | www.acousticmanufacture.com.pl
Acoustic Systems | www.acousticsystems.com
Acoustic Sciences | www.acousticsciences.com
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ATS Acoustics | www.atsacoustics.com
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ClearSonic | www.clearsonic.com
Crossley Acoustics | www.crossleyacoustics.com
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Garvan Acoustic | www.garvanacoustic.com
GeerFab Acoustics | www.geerfab.com
GIK Acoustics | www.gikacoustics.com
Golden Acoustics | www.goldenacoustics.com
G&S Acoustics (Golterman & Sabo) | www.gsacoustics.com
Guilford of Maine | www.guilfordofmaine.com
International Cellulose Corp. (ICC) | www.spray-on.com
Isinac Acoustic World | www.isinac.com
Jocavi Acoustic Panels | www.jocavi.net
Konto Acoustics |www.konto-acoustics.com
Media Specialty Resources - MSR Acoustics | www.msr-inc.com
Norto | www.norto.dk
Offecct | www.offecct.com/acoustics
Overtone Acoustics | www.overtoneacoustics.com
Performance Media Industries, Ltd. | www.pmiltd.com
pinta acoustic | www.pinta-acoustic.com
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VocalBooth | www.vocalbooth.com
Wenger Corp. | www.wengercorp.com
WhisperRoom Sound Isolation Enclosures | www.whisperroom.com

Acoustic Diffusion Research

This article shares Norman Varney's journey on developing new diffuser solutions, and lessons learned while designing countless critical listening rooms around the world. He was actively involved in multiple acoustics standards committees, including for testing procedures, and he explains why diffusers continue to be misunderstood and many times wrongly used.

By
Norman Varney
(AV RoomService)

Norman Varney tests a diffuser around 2018.

Frequency and spatial irregularities are especially problematic in small rooms, such as recording studios and listening rooms. Adding diffusion to the room is almost always beneficial. Acoustic diffusion removes the hard, specular room reflections that interfere with the direct sound of the loudspeakers. They are typically used to control first-order reflections, enhance surround sound envelopment, control flutter echoes, and/or to add more aural openness and sparkle to the space.

There are many diffusers out there from which to choose. I have designed and tested many for my company, AV RoomService, for Kinetics Noise Control, and for Owens Corning. I was, many years back, part of the Audio Engineering Society (AES) SC-04-02 committee, developing standards for acoustical testing of diffusion in labs.

I have learned quite a bit over the years. Probably the most significant takeaway is that diffusion is misapplied by end-users more often than not, therefore, not performing as designed. In developing our new PolyFlex diffuser, our goal was to offer an affordable product that would be applicable for many situations, rather than being specific and therefore limited.

Most diffusers are limited in their applicable bandwidth, effective angles, effective receiver distance, and so forth. Most diffusers are difficult and expensive to make or manufacture. Many are heavy and/or require a lot of real estate. Many of the DIY number theory diffusers end up worthless due to lack of precision in design and/or build.

So how are diffusers wrongly being used so often?

- They are located in the room where they offer little to no effect for the oncoming sound waves.
- They are located in the room where they offer little to no effect for the listener.
- They are often physically too small to be perceived as valuable.
- They are not designed or built correctly.
- The acoustic data is misrepresented.

To begin, let's discuss some of the types of diffusers and their limits. There are geometric-shaped diffusers (e.g., pyramidal, polycylindrical, etc.), and then there are the number-theory diffusers (e.g., primitive root, maximum length sequence, and quadratic residue, such as well and skyline type, etc.). The problems with the cool-looking number theory type of diffusers are where most all of the problems with application arise.

Wavelength Versus Diffuser Size—Does Size Matter?

Size completely matters! Most people don't realize that the size of the diffuser is directly associated with the frequencies it can affect, and what that means. It means that most of the diffusers you encounter for small rooms are only affecting high frequencies. They are likely only affecting maybe the top two octaves on the piano and higher. Be sure to check the manufacturer's spec sheet, including how the test was conducted, which we will discuss.

Let's look at sound wave lengths versus a simple pyramidal diffuser that is one foot square in size.

Note that at low and midrange frequencies, the diffuser is not seen by the sound wave because it is too small compared to the wavelength of the sound wave. At about 1kHz, it does a nice job of scattering. But then at about 10kHz, it behaves specular again (angle of incidence equals angle of reflection). In order to affect lower frequencies, the diffusers must be proportionally large enough to be of influence to the size of the wave. They become unruly very fast. They may even become architectural elements of the room construction design, rather than add-on elements to the room's existing interior.

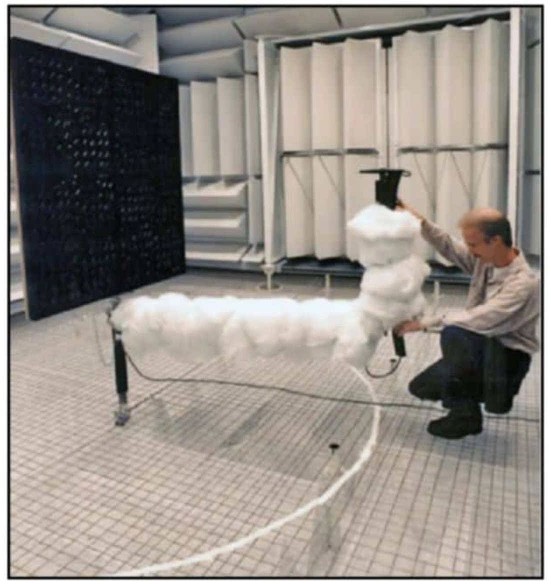
Diffuser/Listener Geometry

Remember that the higher the frequency, the more directional it is and the smaller the wavelength. Think about how many diffusers, especially number-theory (well or skyline) type diffusers, may cast a shadow upon themselves if the sound source and/or listener is off axis. Such diffusers work best behind the listener where the sound incidence and receiver are at normal or zero degrees. They will likely offer little to nothing located anywhere else. In addition, number type diffusers require a distance to the listener before diffusion is effective.

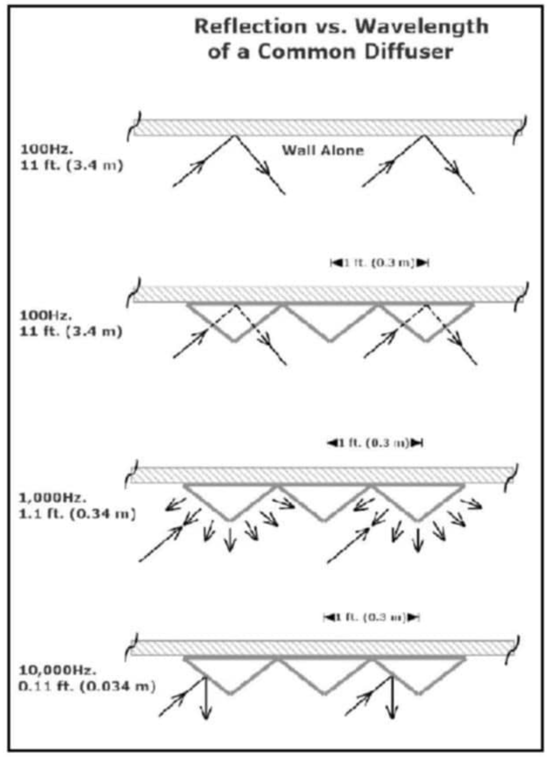
Materials Testing: Diffusion and Scattering

Before we discuss scattering and diffusion we need to understand how the terms are defined in the standards that are used to measure them. Specular reflections are those where the angle of the reflection is always equal to the angle of incidence of the source energy. This is known as Snell's Law.

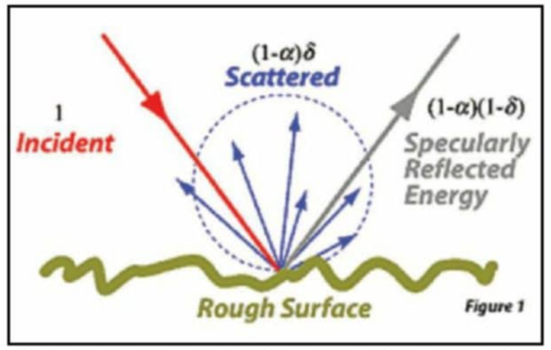
Scattered energy is defined as the energy left over after subtracting specular energy loss from



Norman Varney tests a diffuser around 1998.



Here's a look at sound wave lengths versus a simple pyramidal diffuser that is one foot square in size.



Specular reflections are those where the angle of the reflection is always equal to the angle of incidence of the source energy. This is known as Snell's Law.

total energy loss. Scattering is a measurement of random surface roughness, not structured or large-scale reflectors. The definition removes all surfaces that have a structural depth exceeding 1/16 of the sample size or surface size. Examples would be large geometric reflectors (e.g., pyramidals, cylindricals, etc.), where the constructs are made up of surfaces that have large specular reflection surfaces, and are exempt from this definition.

What Is the "Scattering Coefficient"?

The "Scattering Coefficient" is a parameter that is very much like the "Absorption Coefficient." It describes the loss of energy in a total energy field and is defined as the resultant of subtracting the specular reflected energy from the total reflected energy. The energy loss is then divided by the area of the sample and becomes a "coefficient."

How Is the "Scattering Coefficient" Used?

Simulation programs used for architectural acoustics use coefficients to statistically calculate sound energy left in a room after a defined period of time. Up until the last few years, the programs used only absorption coefficients to calculate the total absorption of energy in a room. Lately there has been an effort to include "scattering" in the calculations to simulate the loss of energy as well as the randomizing of reflections from the surfaces of the "room." The absorption parameter is used to calculate the reduction of energy in the room. Scattering can also be used to do the same thing but is also used to indicate the percentage of energy that is randomly reflected back into the room versus the energy reflected in a specular way.

Absorption had well known test procedures that were later standardized to create the parameter, "absorption coefficient," but scattering coefficients were inserted into the programs and "tweaked" by users based on "experience." This method was better than nothing as long as measurements were not available and standards had not been developed. This ended with the adoption of ISO-17497-1.

The standard, ISO 17497-1, was developed in Europe. It was based, primarily, on research conducted by Michael Vorländer, Institute of Technical Acoustics, and Eckard Mommertz, Müller-BBM GmbH, using a free-field method. This was later converted to a method that used a reverberation room instead. Initially all research and measurements were done using scale models of the scattering surface. NWAA Labs was the first research and testing facility to develop a "FULL SCALE" method to perform scattering tests. There were numerous initial problems converting the scale model method to full scale, but Ron Sauro made breakthroughs developing "corrected" procedures to get consistent and repeatable measurements of the full scale samples.

Customers came to NWAA for full-scale measurements, and using reverberation rooms of medium volume, results were obtained. Many customers did not really understand the data, but accepted it. Even though NWAA Labs normally uses the standard test procedures described for testing, Ron Sauro is constantly trying to improve or create better ways of measuring relevant acoustic parameters.

After moving the laboratory from California to the present location in Elma, WA, Sauro set up the same exact system in the much larger facilities intended to measure low frequencies in a more accurate way. They installed the identical equipment, software, and procedures and the same samples in the new facilities and proceeded to run the exact same tests.

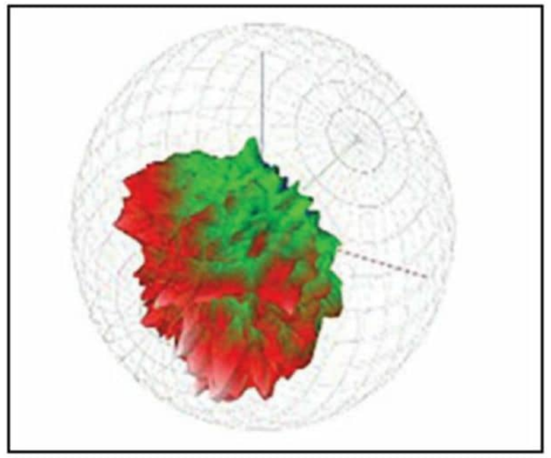
They had already qualified the new reverberation rooms using ASTM and ISO standards. The results from the new tests did not confirm the tests previously performed at the other facility. They performed the tests again and again, and each

Ron Sauro stands next to his speaker/diffuser test rig at NWAA Labs.

Resources

AV RoomService, Ltd., https://avroomservice.com

M. Vorländer and E. Mommertz, "Definition and Measurement of Random-Incidence Scattering Coefficients," SCRIBD, www.scribd.com/document/111726956/Vorlaender-Mommertz-Scattering



A complimentary section of the standard allowed a 180-degree by 360-degree measurement of an in-wall mounting of a speaker. This results in a hemisphere of the front facing half of a directivity balloon shown here as a balloon diffusion 3D plot: frequency, amplitude, and angle.

time the data returned was completely random. This forced Sauro to go back into a research mode and find out why he was not able to get answers that matched the previous data.

They built another reverberation room (temporary) of the same volume as the old testing facility. The equipment and samples were then reinstalled and the testing was done again. The answers came back close to the original data. The difference was the size of the reverberation room! The difference was that larger rooms were more diffuse than the smaller reverberation room. This pointed to an error in the current standard, the assumption of diffusivity in the smaller rooms.

A truly diffuse room has an energy field that has a fully "mixed" energy magnitude, and energy phase, everywhere in the room (away from the boundaries).

The idea of comparing phase differences to remove energy from the energy field cannot be accomplished when you cannot have differences in phase responses from different impulse responses in a fully "mixed" phase field. You can detect differences in a free field but not in a truly diffuse field. This means the "standard" for measuring scattering cannot operate the way it was described and can only operate in a non diffuse field. As a full scale measuring standard, it is not a useable way to measure this parameter.

Ron Sauro has developed a new testing regime for diffusion testing that provides full 360-degree balloon data as well as polar, frequency response, phase response, and magnitude data. This regime is now the basis for a new standard being developed by ASTM for the testing of diffusion.

How to provide a useable parameter to work inside of simulation programs is the problem. The solution was to look into the past and adapt what we already know. ASTM started looking, with guidance from Sauro, into what the function of "diffusion" panels and structures are. Sauro thought that diffusers perform exactly the same task as speakers do. They take energy from a source and radiate this energy into the room with a directivity function.

Speakers provide their own energy in using a voice coil to convert electrical energy to acoustic energy and using the cone or horn to "direct" the energy. Diffusers do the same thing using reflected energy. How do we use this information in the simulation programs? We provide "directivity balloons" at each one-third octave frequency from 100Hz to 10kHz at each 5-degree resolution (19 calibrated microphones on an arc) around the front of the speaker, for in wall speakers. We can provide the same "reflection balloon" for various reflection devices? Enter the development of a new standard!

Developing a New Standard at ASTM on Surface Reflectivity

Sauro had measured speakers for many years using the AES-156 standard. This standard developed a directivity balloon of the sound radiation in 5-degree increments using a matrix of a 360-degree by 360-degree balloon matrix. A complimentary section of the standard allowed a 180-degree by 360-degree measurement of an in-wall mounting of a speaker. This results in a hemisphere of the front-facing half of a directivity balloon being shown.

This allowed the use of speakers in simulation programs for acousticians, such as EASE, Odeon and CATT Acoustics. Sauro thought this might be a great way of including reflectors and diffusers in

About the Author

Norman Varney is the principal consultant for AV RoomService, Ltd., established in 2001. During that time, Norman was also Acoustical Product Development Manager for Kinetics Noise Control, and prior to AVRS, was with Owens Corning at the Science & Technology Center, where he was a Senior Engineer with the Acoustic Systems Business as the Acoustic Design Center Lead. Prior to Owens Corning, he worked at Music Interface Technologies where he designed critical listening and viewing environments, AC line conditioners, video cables, and was Director of the Custom Installation and Home Theater divisions. While there, he designed the very innovative and elaborate electrical system for the Scoring Stage of Lucas Film's Skywalker Ranch and was the lead for the 2C3D and 5C3D Certification Programs. Before that, Norman owned Alta Buena Stereo, a retail store in California for 12 years.

He has designed more than 500 critical listening rooms around the world and is an active member of American Society for Testing and Materials (Committee E-33 on Environmental Acoustics), Acoustic Society of America, Institute of Noise Control Engineering, Audio Engineering Society (sub committee AES-4id-2001 for establishing diffusion coefficient standards), National Association of Music Merchants, Custom Electronic Design and Installation Association, and others. He has written and presented white papers for the industry and continues to have articles published around the world.

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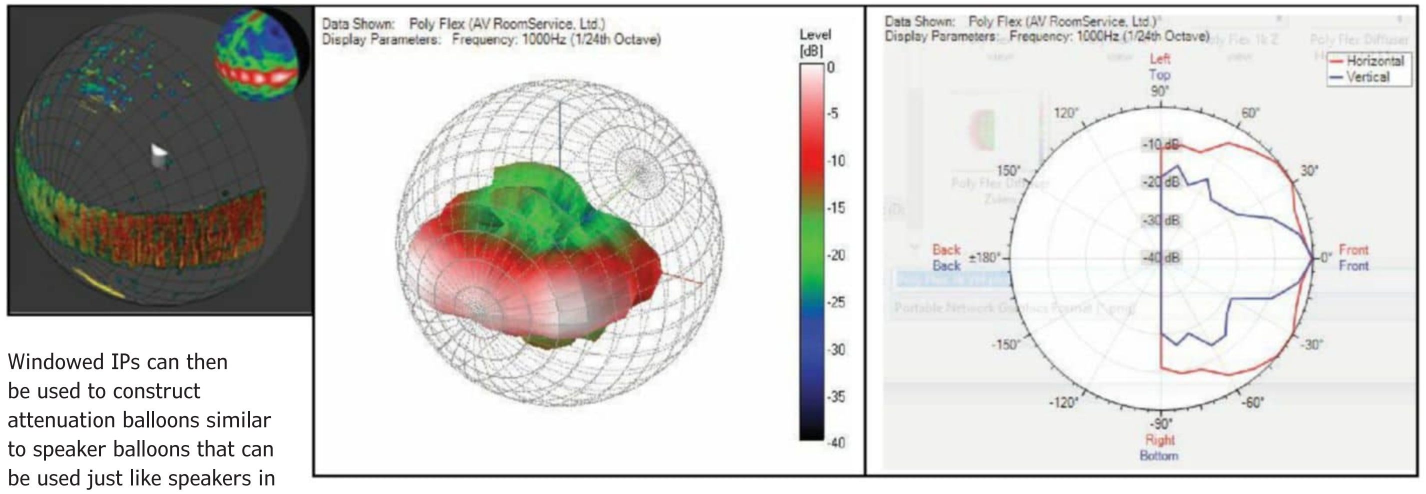

Windowed IPs can then be used to construct attenuation balloons similar to speaker balloons that can be used just like speakers in the simulations programs. Shown here is a balloon comparison: simulated vs. measured vs. typical polar plot.

the simulation programs, as well. The question was how to provide a signal to be measured from these reflectors! Speakers have a voice coil to convert an electrical signal to acoustic energy by moving a diaphragm. This makes the air compress and decompress to move the eardrum. Reflectors do not have self-contained "voice coils." They are intended to "reflect" energy already present in the "room."

In ISO-17497-2, a source was proposed outboard of the measuring system so that the direct sound can be "windowed" out of the Impulse response. This results in the measurement of first-order reflections from the "diffuser" or DUT.

These windowed IPs can then be used to construct attenuation balloons similar to speaker balloons that can be used just like speakers in the simulations programs. With such proven testing means more confident modeling programs, and therefore better designed diffusers.

The PolyFlex diffuser has polycylindrical shape with an opened end, and we made it flexible.

Going back to our design criteria for best-bang-for-the-buck, we selected the polycylindrical shape with an opened end, and made it flexible. This meant the PolyFlex diffuser offered the following attributes:

- Wide bandwidth (630Hz on up)
- Wide effective receive and return propagation angles (150 degrees)
- No receiver distance requirement (no convergence point)
- No temporal time (phase) skewing
- Tunable frequency and propagation to prevent lobing
- Can act as an absorber too by adding material behind
- Can be oriented for either vertical or horizontal scattering

The PolyFlex diffuser is a 1D diffuser, meaning it will scatter either horizontally or vertically, depending on orientation. It is available in a 100% clear acrylic or a neutral grey metal. They can be placed behind stretch-fabric systems or painted.

PolyFlex diffusers can add sonic spaciousness to the room, air around the instruments, improve timbre accuracy and/or widen the soundstage. They also improve clarity in speech and spatial focus without making the room sound dead and/or small. PolyFlex diffusers can also be used at reflection points in place of typical absorbers when room reverberation times are already low. Once the reverberation times of the room are under control, it would be very difficult to have too much diffusion added to the room. We feel that the PolyFlex met our design criterias in spades!

Author's Note: I would like to thank Ron Sauro for his insight and work, which has provided the advancement of diffusion understanding, testing, and design for all to benefit.

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Reverberation: Control and Measurement

Since the construction of the first performance venues, acousticians have struggled with acoustical interference, reverberation, and echoes. In this article, our acoustics expert discusses ways to control sound and how to accurately measure it.

Photo 1: This ancient Greek amphitheater includes stone seats sloping downward toward the stage.

By
Richard Honeycutt

The first performance venues we knew about were threshing floors surrounded by gentle upward sloping fields upon which people would sit or stand. Ancient amphitheaters (**Photo 1**) were developed from these structures. With the introduction of enclosed venues (temples, and later, performance halls), reverberation became a topic of concern. Marcus Vitruvius Pollio, a first-century Roman architect, in Book 5 of his "Ten Books on Architecture," discusses such acoustical phenomena as "dissonance," which we would call acoustical interference; "circumsonance" (reverberation); and "resonance" (echoes). Reverberation is the sound that bounces around in a room after the initial sound.

Photo 2: Harvard University's Fogg Lecture Hall was the site of Wallace Sabine's fundamental research on reverberation.

During the 19th century, efforts toward acoustic enhancement of auditoria and concert halls were principally directed toward providing reflecting surfaces (sounding boards) to concentrate the energy from a speaker, a singer, or instrument(s) toward the audience. In 1853, Dr. J. B. Upham of Boston, MA, undertook a careful study of the acoustics of the Boston Music Hall, culminating in an essay on reverberation and resonance.

American physicist Joseph Henry presented papers to the American Association for the Advancement of Science in 1854 and 1856, and a new lecture room in the Smithsonian Institution was built according to conclusions of his experiments and theory. The results were said to be highly satisfactory. Architect T. Roger Smith published his book *Acoustics of Public Buildings* in 1861, in which he discussed the differing acoustical requirements of speech and music. John Tyndall and John William Strutt (Lord Rayleigh) investigated the control of reverberation in rooms.

W. C. Sabine's Groundbreaking Work

But the definitive work that is universally agreed to mark the genesis of scientific acoustics was that of Harvard University Physics Professor Wallace Clement Sabine. In the very last years of the 19th century, Sabine was tasked with remedying the horrific acoustics of the Fogg Lecture Hall (**Photo 2**). He began with a quantitative study of the nature of reverberation and the physical factors controlling it. He concluded that the poor speech intelligibility in Fogg Hall resulted from excessive reverberation. He developed a metric for reverberation in a room, which he called Reverberation Time (RT), and he defined it as the time required for the sound to decay by 60dB after the cessation of the direct sound from the source. Recognizing that reverberation control involved acoustical absorption, which varied with frequency, Sabine measured the RT at different frequencies using a stopwatch. The sound source was an organ pipe tuned to the frequency of interest, which he blew himself (**Photo 3**). Dividing the total acoustical absorption by the area of the absorbing material, he defined an acoustical absorption coefficient α, which he described as the fraction of incident sound energy absorbed by the material. This led to his famous "Sabine equation" for RT:

$$RT = \frac{0.049V}{A}$$

where *RT*= reverberation time,
V= the volume of the room in Imperial units,
and *A*= the total acoustical absorption in the room.

A, quantified in sabins, (traditionally defined as the acoustical absorption of one square foot of a perfect absorber) is the sum of the products of the area *S* of each material and the α of that material:

$$A = \sum_{\eta=0}^{N} S_\eta \alpha_\eta$$

As discussed in the April 2016 and December 2017 Sound Control articles, acousticians have noted that Sabine's description of α as the fraction of incident sound that is absorbed is not quite accurate: an α of unity does not give zero RT, although if α were indeed the fraction of incident sound absorbed by a material, having an α of unity would leave no sound in the reverberant field, and thus zero RT. Therefore, we must define α as a constant of a particular material which, when obtained using the standard measurement techniques and the Sabine equation, allows prediction of RT with some accuracy, given certain known limitations. This new definition allows the possibility of $\alpha > 1$, which is what is observed when some materials are measured using the standard testing procedures.

Sabine's equation was developed with the assumption that reverberant sound waves are random in direction, implying a perfectly diffuse room. Because of non-uniform placement of absorption, acoustical focusing by reflective surfaces, and other artifacts, real rooms always deviate somewhat from perfect diffusivity; thus some error in RT prediction by Sabine's method is to be expected.

Listener Perception of RT

Since RT was the first architectural acoustical parameter to be quantified, it became the focus of architectural acoustics. The effects of RT variation with frequency were studied. Since this variation results from every material's α depending somewhat upon frequency, and the dependence manifesting more as a general trend than as marked discontinuities, graphs of RT versus frequency were usually plotted in octave bands; narrower frequency bands were unnecessary.

For lecture halls and other speech rooms, an RT characteristic that is flat with frequency is preferred, as it yields the best speech intelligibility. But for music performance spaces, a slight boost at the lower frequencies enhances the sense of musical balance; rooms having no such low frequency RT boost tend to sound thin. A gently sloping

Photo 3: These wooden organ pipes are similar to the ones Wallace Sabine used as sound sources in his RT measurements.

About the Author

Dr. Richard Honeycutt fell in love with acoustics after his father brought home a copy of Leo Beranek's landmark text on the subject when Richard was in the ninth grade. Richard is a member of the North Carolina chapter of the Acoustical Society of America. Richard has his own business involving musical instruments and sound systems. He has been an active acoustics consultant since he received his PhD in electroacoustics from the Union Institute in 2004. Richard's work includes architectural acoustics, sound system design, and community noise analysis.

high-frequency decrease in RT prevents a room from sounding harsh. Probably humans' preference for this RT signature comes from the fact that most room finishes, and especially upholstery and audiences, have higher absorption as frequency rises. In any case, rooms in which the RT in the

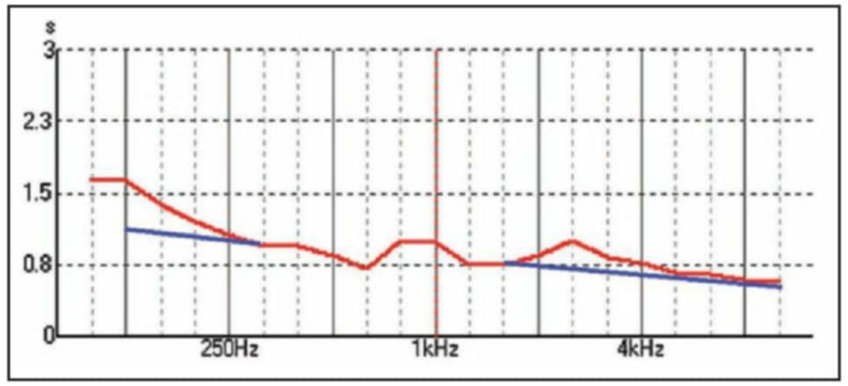

Figure 1: The red curve shows the actual variation of RT with frequency in this auditorium; the blue lines inicate the preferred maximum bass rise at low frequency and treble droop at high frequency.

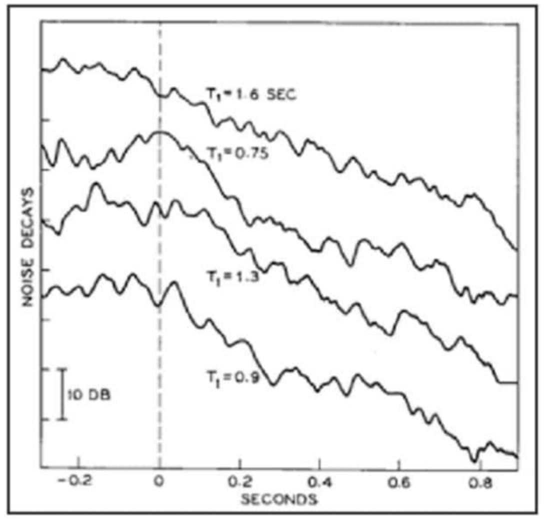

Figure 2: These four graphic level recorder traces show the sound decay in a large concert hall, measured under different conditions.

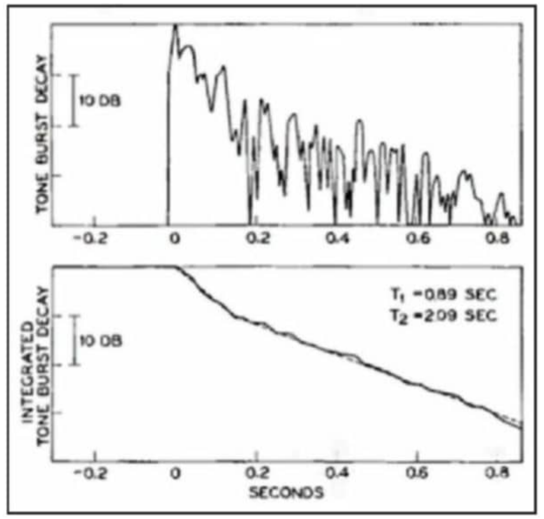

Figure 3: The top graph shows the squared sound-burst signal, and the bottom one shows the result after integration.

125Hz octave band exceeds 140% of the midband (500Hz) value, the room will sound boomy. If the RT in octave bands above 2kHz is less than 80% to 85% of the midband value, the room sound will be perceived as muffled or "blanketed."

Figure 1 shows the RT versus frequency for a medium-sized university auditorium, with limits added showing the preferred maximum low-frequency boost and high-frequency droop. This auditorium, although not ideal in terms of the RT signature, is tolerable. The RT in the lowest octave is abut 160% of that in the midband, making the room sound somewhat bass-heavy.

After decades of considering the full RT of listening rooms, researchers determined that the early part of the reverberant decay is more important than full RT in terms of audience perception. The time required for the sound level to decay 10dB from maximum is more predictive of listener preference for a room than is the full RT. This Early Decay Time (EDT) leads a listener to conclude whether the room "sounds right" or not.

Measuring RT

Although stop watches were still being used to measure RT as late as the early 1980s, organ pipes were no longer used as the sound source. Because of the gradual change in a material's sound absorption with frequency, the narrow bandwidth provided by an organ pipe provided no advantage. For average—rather than frequency-specific—RT measurements, impulsive sounds such as balloon bursts or shots from a starter pistol were used. Once electronic signal sources had become sufficiently developed, octave-band-filtered white or pink noise sources became common. In this case, a microphone and preamplifier feeding a graphic level recorder provided the record of the signal decay (**Figure 2**) [1].

In 1965, Manfred Schroeder of the Bell Telephone Laboratories introduced a new method of measuring RT. He initiated a white-noise sound burst, waited long enough for the sound level in the room to reach steady-state (in practice, about 2 seconds), and used a tape recorder to record the decay of the sound. Then he played back the recording in reverse. (With a full-track mono recorder, this was easy to do.) The time-reversed signal was then squared to eliminate negative values, and integrated using a single-pole RC integrator (low-pass filter). The top graph of **Figure 3** [2] shows the squared noise-burst signal (the impulse response), and the bottom graph shows the integrated result. Schroeder's method provides a much smoother decay curve than a graphic level recorder trace, leading to more consistent RT measurements.

Today, most RT measurements use a modified form of Schroeder's method in which the tape recorder is replaced by a digital recorder, and the squaring and reverse integration are performed by computation rather than by a passive electrical circuit. **Figure 4** shows the results of such a measurement. The curving below 60dB shows intrusion of background noise into the impulse response. There is also nonlinearity above 90dB, due to strong early reflections that are not considered a part of the reverberant decay. Clearly, the decay has no single slope except for the region between 60dB and 90dB, which is almost a straight line. Therefore, current standard practice is to eliminate the first 5dB of decay, then multiply the time *t* required for the quasi-linear decay *d* by 60/*d* to get the RT.

In this case, the quasi-linear decay *d* is 30dB, and the time required for the Schroeder curve to drop from 89dB (5dB below peak) to 59dB is about 0.61 seconds. Since this represents a decay of 30dB, 60/*d*=2, so we multiply 0.61 seconds by 2 and get RT=1.22 seconds. To specify how much decay we actually measured (30dB), we call this value T30. In a noisier room, we might have to make a T15 or T20 measurement. While T10, T20, and T30 are seldom exactly the same, they are usually pretty close in a reasonably diffuse room. But they are virtually always higher than the EDT.

To help clarify the effect of Schroeder reverse integration, look at **Figure 5**, which shows the result of a computer simulation of a different auditorium. Note that starting from the right, the blue Schroeder curve shows a value of zero, because there is no sound energy until the cluster of reflections shown before 1000ms. Continuing to move from right to left (backward in time), we see that as each significant group of reflections occurs, the Schroeder curve increases. As the reflections become denser (closer together in time), the Schroeder curve smooths out. The section used in computing T30 is really not very smooth, which would result in variations among the values of EDT, T10, T15, T20, and T40. The reason that the impulse response is so ragged is that there is not enough energy in the signal source, making some groups of reflections "stand out from the crowd."

In the simulation, using more rays (or "particles," in EASE terminology) would give us a smoother curve. When actually measuring RT, our choice of signal source provides the same effect. Impulse responses from handclaps, balloon bursts, or starter pistols have long been used for RT measurements. Handclaps have little energy; balloon bursts have more; and starter pistols, still more. But the low-frequency energy of any impulse source is low. Thus most instruments use noise signals—white, pink, or Maximum-Length Sequences (MLS), or swept-frequency sine-waves. These have been listed in order of increasing energy; thus, increasing desirability. However, the computational load in using non-impulse sources is higher than for impulsive sources. Fortunately, the instrument-makers' programmers handle those consequences for us!

We will examine some RT-measuring instruments in a future Sound Control article.

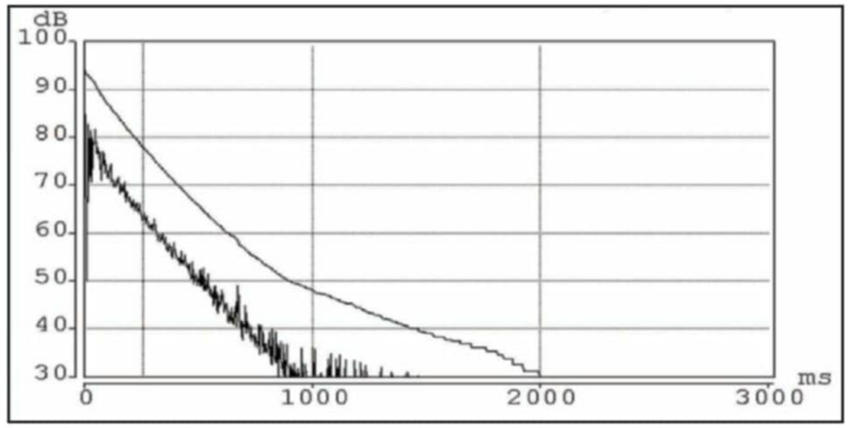

Figure 4: The digitally recorded impulse response and Schroeder curve provide more repeatable results than older methods.

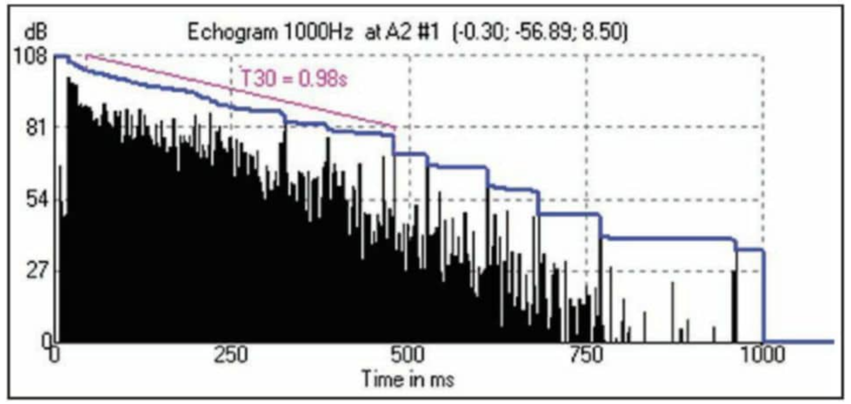

Figure 5: This shows a sample T30 of a different auditorium, produced by a computer simulation.

References

[1] M. R. Schroeder, "New Method of Measuring Reverberation Time," *Journal of the Acoustical Society of America* (JASA) 37, pp. 409-412, 1965.

[2] Ibid.

Distributed Subwoofers for In-Room Low-Bass Problems

After discovering room modes in his listening room, Thomas Perazella opted to solve his problem by building and placing multiple subwoofers at different locations in the room to create a greater distribution of modes.

By
Thomas Perazella

In the August 2021 issue of *audioXpress*, I detailed my project using acoustic treatments to tame problems in my dedicated audio room. In that room, the speakers were placed in the optimum positions for the mid and high frequencies. Inevitably, that positioning leads to problems with bass frequencies due to speaker boundary interface problems that cannot be effectively ameliorated with the room treatments I used even though they were quite successful for the mids and highs. Also, with nulls at the offending frequencies, trying to eliminate them electronically is usually a fool's errand.

If there is a cancellation at a given frequency due to the relative positions of the speakers, boundaries, and listening position, putting more power into the speakers at that frequency will not only increase the acoustic output from the speakers, but also the reflected sound that is causing the cancellation. It may not be a zero-sum proposition, but the power required inevitably leads to higher distortion and power loss along with excessive bass in other parts of the room, increasing the chance of annoying vibrations from room contents.

Schroeder Frequency

With rooms, there is a frequency commonly called the Schroeder Frequency named after Manfred R. Schroeder, usually in the range of 100Hz to 200 Hz for small rooms where, above that frequency the sound generated by the speaker and room interfaces has so many reflections that they tend to become diffuse with no major modes. Below that frequency, the relationship of the speaker positions to the room boundaries causes fewer resonances, which can either reinforce or cancel those frequencies depending on the locations and frequencies. Having fewer modes leads to significant response variations.

Contrary to some common beliefs, trying to eliminate the room modes is not a reasonable solution due to the difficulty of modifying low-frequency reflections in a room. Rather, it has become more popular to increase the modes to average the response in the room, not dissimilar to what happens above the Schroeder Frequency. This is done by having multiple sources of the low frequencies in different room locations. Much history and information about this technique is available on the Internet.

"Sweet Spot" vs. Multiple Locations

Before embarking on a project to minimize the offending modes, it is important to decide whether you will have one "sweet spot" for individual listening or a broader requirement of multiple locations common to a home-theater application. Although minimizing the low-frequency fluctuations in a room is an involved procedure, it is more simple for one location. In my case, I have only one prime location so I was able to use a simpler approach. This also made the locations of the subs more conducive to fitting into other room considerations since the audio room serves double duty as my office.

Primary Problem

Although my audio room is rectangular with the speakers equidistant from boundaries and a listening position centered between the speakers, there are differences in the responses at the listening position from the left and right speakers at low frequencies. The left wall has no windows but a door on the rear right at the corner. The right side has two windows toward the front and rear corners. In addition, the left wall faces into the house and has a layer of Homasote sound absorbing material beneath the drywall. The right wall faces the outside and has only drywall. Both walls have fiberglass insulation in the spaces between the wall studs.

Figure 1 shows the response at the listening position when only the left woofers are in play. You can see there is a significant suck out of nearly 30dB at just under 40Hz. **Figure 2** shows a different response from the right woofers only with a dip at nearly the same frequency but only about 7dB deep. These differences necessitated treating the two channels separately. **Figure 3** shows the composite response with both channels operating. The suck out is reduced to about 22dB but is a little broader.

Solution

If you put multiple subwoofers at different locations in a room you can create a greater distribution of modes. If the subs are properly located and time aligned, a smoother bass response at the listening position can result because of averaging of the various modes, both reinforcing and cancelling. This is the process I decided to use. Research has shown that from a cost/performance perspective four subs are optimum. Therefore, I decided to use four subs, but in an unconventional configuration.

Since the listening room does double duty as my office, when furniture was placed in the room, consideration of both functions had to be taken into account. The room is 20′ long by 15.5′ wide by 9′ high. Looking at the room from an audio perspective, the front wall behind the speakers is 15.5′ wide. Centered on that wall is a window. The two speakers are placed out 5′ from the front wall and 4′ from the side walls. There is an "L" shaped couch placed so that the prime listening position is centered between the speakers and 5′ in from the rear wall. Behind that couch and facing the right wall is my desk, which is "L" shaped and also a separate set of drawers. The right side of the couch "L" is spaced 12" from the right wall.

Given these constraints, I decided to put two of the subs in the space between the right side of the couch and the other two behind the rear section of the couch. Two factors were key in the design of the subs. First, the 12" space between the right side of the couch and the right wall necessitated a narrow

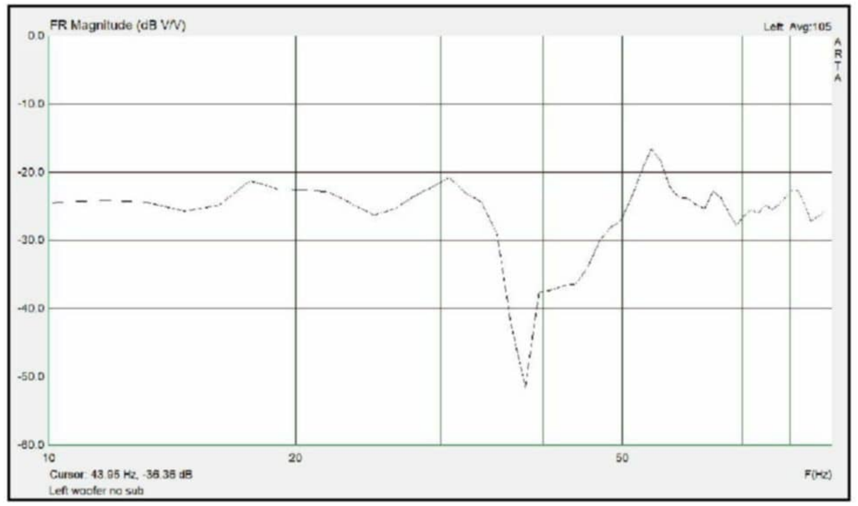

Figure 1: This listening position response was measured with left woofers only.

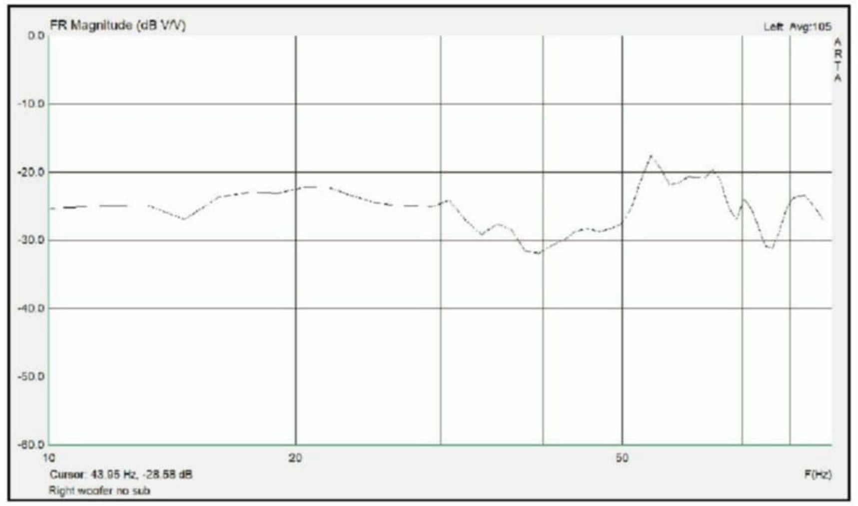

Figure 2: This shows a different response taken using the right woofers only.

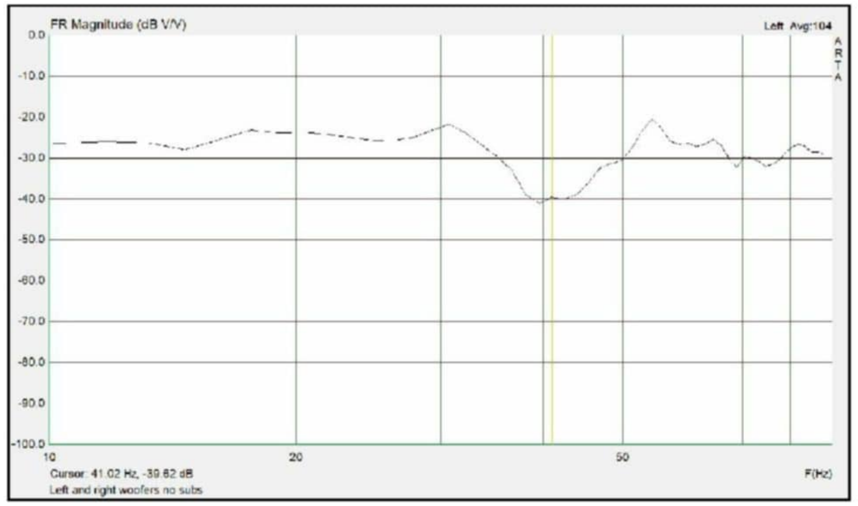

Figure 3: This listening position response was taken using both the right and left woofers.

enclosure design. The second set of subs would be placed directly behind the rear side of the couch with one driver directly behind the listening position. To reduce the potential of enclosure movement from such a narrow enclosure, I decided to create two long narrow cabinets with a driver mounted to each small end. That way the vibrations from the drivers would be turned into tension and compression forces on the cabinets, which would effectively cancel them out, resulting in less movement even at large excursions. I have done this many times in the past and the results were quite good.

The Drivers

Given the restricted space between the couch and the wall, I felt that the maximum diameter for the drivers would be 10″. I was not concerned with that limitation because the two main bass cabinets for the speakers each use two high excursion 12″ drivers that had sufficient linear volume displacement to handle the lowest frequencies. The main function of the distributed subs was to fill in the suck out at 40Hz and they did not need to plumb subterranean frequencies. Fortunately, we are blessed with many high-quality subwoofer drivers at prices that offer great value. Researching the choices available, I settled on the Dayton Audio RSS265HF-8 (**Photo 1**).

It is a 10″ driver that has an amazing 14.3mm Xmax. Additional information about this driver can be found on the Parts Express website (www.parts-express.com). The woofers in the main speakers are the Dayton Audio RSS315HO-44. I have had great success with them so I felt confident that these HF Series drivers would work equally as well. The model I selected had an impedance of 8Ω so that using two in parallel resulted in a reasonable 4Ω impedance.

Photo 1: I used the Dayton Audio RSS265HF-8 10″ driver in the subs.

Cabinet Design

The material chosen for the cabinet walls was ¾″ MDF, which is the material I have used for years in my speaker projects. Looking at the drivers' Thiele-Small (T-S) parameters, I decided that the optimum volume for each driver was 2.5ft³. The outside dimensions of the dual enclosure were 47.5″ long by 12″ wide by 22 3/8″ high. Given the thickness of the MDF, that resulted in about 2.8ft³ for each driver, giving a calculated fB of 32Hz and a QB of 0.63. Some of that volume would be taken up by the parts of the driver inside the enclosure but the apparent volume seen by the driver would be increased by the stuffing used. I employed my usual construction method when using MDF, that is biscuits and wood glue. When butt joining MDF panels, biscuits greatly reinforce the joints.

The MDF was sourced from a nearby hardware store. Two 4′ × 8′ sheets were required. After laying out the panels, I decided that for ease of transport, the wood sheets could be cut in half at the store to make four 4′ × 4′ panels. **Figure 4** shows the actual dimensions of the panels. The overall dimensions of the end panels are the same but one end has a cut-out for a Speakon connector cup at the bottom with the driver mounting hole directly above it.

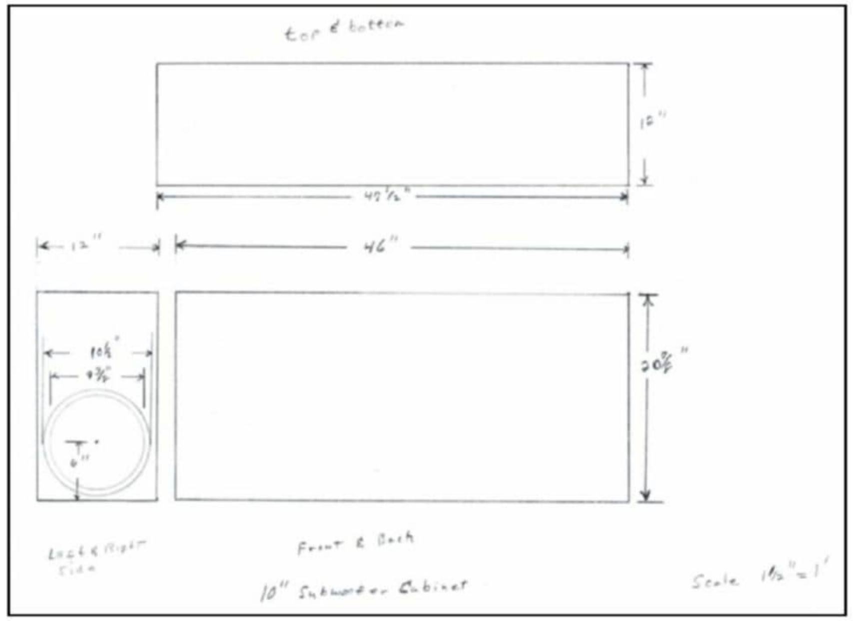

Figure 4: This drawing shows the actual dimensions of the panels.

Photo 2: I used this stack of cut panel pieces to construct the cabinets.

Construction

The 4′ × 4′ panels were cut to the sizes of the individual pieces on my table saw. **Photo 2** shows the stack of cut pieces. **Photo 3** shows the Speakon connector cup.

When working with ¾″ MDF driver mounting panels I sometimes use two sheets laminated together to provide additional rigidity and a thicker depth for the mounting screws. In this case, I would

Photo 3: This is the Speakon connector cup that I used.

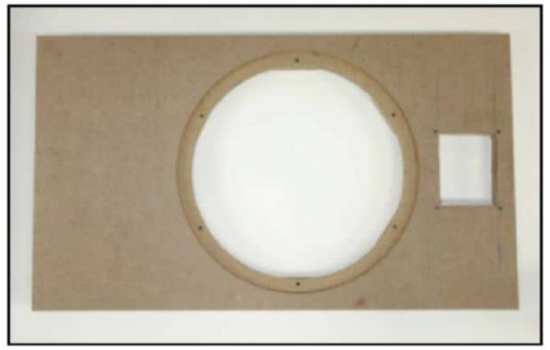
Photo 4: This is the outside mounting panel with driver and Speakon cup holes.

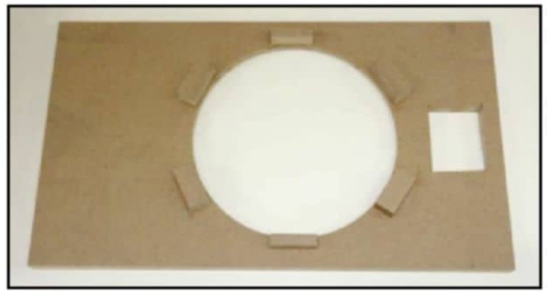

Photo 5: The rear side of mounting panel with the backing pieces is shown here.

Photo 6: The first assembly step entailed gluing the center divider to the side piece.

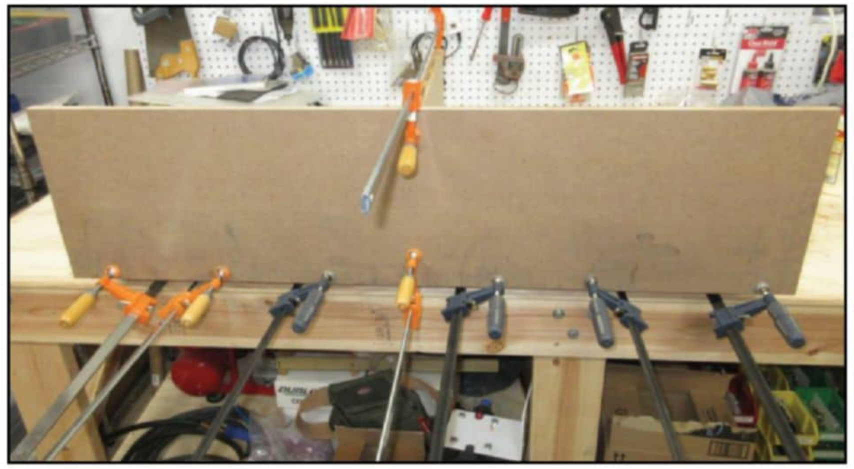

Photo 7: I glued the bottom of the cabinet to the side and the divider.

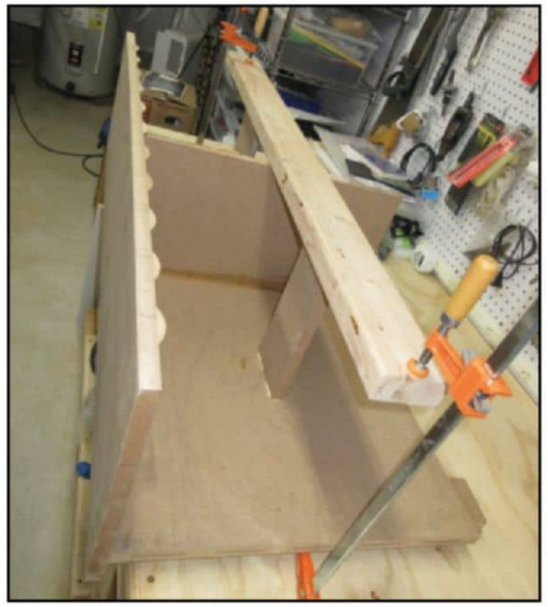

Photo 8: Next, I glued the cross braces to the side panel.

be flush mounting the drivers, which meant the panel thickness would be reduced. The panels were narrow enough that they didn't need greater rigidity. Only one thickness of sheet was required, but I used scrap pieces of MDF glued behind each mounting hole to provide additional thickness so the screws would not pull out.

Photo 4 shows the outside of one of the driver mounting panels with the recess for the driver basket and the hole for the connector cup. **Photo 5** shows the rear side of the panel with the extra pieces of MDF glued behind the screw holes.

Before assembly started, all the necessary slots for the biscuits were cut using a plate joiner. To minimize standing waves in the cabinet and allow for changes in the phasing of the drivers if needed, I made a dividing panel to be placed in the center of the long dimension of the cabinet. The cabinet then essentially became two cabinets joined at one end each. As previously mentioned, if the application was home theater, I would have made four separate cabinets to provide maximum flexibility of placement. **Photo 6** shows the first assembly step, which was gluing the center divider to one side of the cabinet.

Next, I attached the bottom of the cabinet. **Photo 7** shows the bottom clamped to both the side and divider. The panels most subject to vibrational forces would be the two sides. To minimize the vibrations, I glued a cross piece of MDF between those sides to again convert any forces that would tend to bend the panels into tension and compression forces in the cross member.

Those cross members were placed off center on the panels to divide the remaining sections of the panels into unequal sizes, further reducing the potential for vibrations at a given frequency. **Photo 8** shows the cross members being glued using a 2′ × 4′ member to span the whole distance. Looking at Photo 8, you can also see the biscuits in the slots of the bottom in preparation for gluing the other side panel. The side panel was then glued in a similar fashion.

Next, I glued the front and rear driver mounting panels in place. I did not have clamps long enough to span the full length of the cabinet so I improvised. Two scrap pieces of wood were clamped vertically along the side of the cabinet at a distance that would allow the distance between the wood and the end of the cabinet to be reached by some of my long clamps. Then I used that vertical piece to be the base for the clamps holding the driver panels. **Photo 9** shows that setup. In that photo, the old woodworker's saying "Clamp are like dollars, you can never have too many" is very apparent. Gluing the top in place was the last wood assembly step.

Finishing the Cabinets

Regardless of the finishing method you choose for a woodworking project, prep is paramount. I examined the cabinets and filled any gaps or dings before sanding. I also routed the edges with a ¼″ round over bit because sharp edges on MDF are very prone to damage. Sanding was done with a half-sheet sander. Note that in **Photo 10** the sander has a vacuum hose attached. Dust removal is very important to your health with any woodworking project but especially so when working with MDF.

Here is a helpful hint when working with a dust collection system with ungrounded double insulated tools: If you have a metal tool surface that you will contact, it helps to add a grounding wire to that surface to remove the static charge that can build up to high levels due to the dust passing through the collection hose. It is similar to dust particles rising up through a cloud that cause lightning. Without that grounding wire, after prolonged sanding a surprisingly strong shock can result if you touch a grounded item.

My original thought was to apply Formica sheets to the cabinets as a finishing method. I decided since I had no experience with laminates and the cost of two 4′ × 8′ sheets was high, I would choose something else. Painting was an option but trying to match the finish on the main speakers, which I had done by an auto shop was not possible at home. Since the subs would be sitting on the floor with one of them directly behind my office chair where it might get hit by my chair or a vacuum cleaner, I decided on a more durable finish.

Parts Express has a material called Exohyde that looked very durable and also has the ability to produce a textured finish that could hide any minor blemishes in the wood. I got a gallon of that along with a special roller that is used to produce the highly textured finish. Two advantages of working with Exohyde is that no primer was needed and clean-up could be done with just water and soap. Working with rollers, especially those with deep knaps can result in paint spatter traveling a substantial distance from the roller.

When painting the cabinets, I put a large drop cloth on my basement floor and raised the cabinets off the floor using two one gallon paint cans under each. The sequence of painting was to first paint the bottoms and let them dry. Although most finishes give suggestions for drying time, I usually allow a few days for the first sections to dry so that when those surfaces are turned over and placed on the supports, the surface is hard enough to prevent damage. Then the cabinets were turned over on to their bottoms and the balance of the surfaces painted. **Photo 11** shows the painted cabinets.

Photo 9: Here is the clamp range extender setup that I used.

Photo 10: I attached a vacuum hose to the sander to remove all the dust.

Photo 11: I then painted the cabinets with Exohyde.

About the Author

Thomas Perazella is a retired Director of IT. He received a Bachelor of Science degree from the University of California, Berkeley campus. He is a Past President and Treasurer of the Rockville Chapter of the Izaak Walton League of America, one of the oldest national conservation organizations in the US. Audio has been his passion for more than 50 years and he is a member of the Audio Engineering Society, the Boston Audio Society, the Philadelphia Area Audio Group, the DC HiFi Group, and the DC Audio DIY Group. He has written for *Speaker Builder* and *audioXpress* magazines. He has also authored several articles in professional audio journals and taught commercial lighting at the Winona School of Photography. Recently he received a patent on a cost-effective high-efficiency LED lighting system for commercial and residential buildings.

Photo 12: I placed furniture glides on the bottom of the cabinets to make it easier to move them into place.

The weight of the finished cabinets with the two drivers mounted is quite substantial. To facilitate moving the cabinets on the Berber rug in my audio room, I added four large furniture glides to the bottom of each enclosure. **Photo 12** shows those glides. The last step before bringing the cabinets out of my workshop, up to the listening room, and mounting the drivers was to do the internal wiring from the Speakon connectors to the driver locations and then add stuffing. I placed one 5 lb. bag of Acousta-stuf in each cabinet. The wiring and stuffing can be seen in **Photo 13**.

Once the cabinets were in the audio room, I attached the speaker wires to the drivers and fastened the drivers using black square drive screws. One cabinet called Right Sub was positioned between the right side of the couch and the wall. **Photo 14** shows that cabinet in position. The other cabinet called Left Sub was positioned behind the left side of the couch with one driver directly behind the listening position. **Photo 15** shows that cabinet.

Photo 13: The wiring and stuffing is shown inside the cabinet.

Photo 14: I placed the Right side sub between the couch and the wall.

Photo 15: I positioned the Left side sub behind the left side of the couch.

Electronics and Connections

Except for my turntable, preamp, and control panel, all the electronics for the system are on a rack in the basement under the audio room. During the house's construction I had the builder put in four raceways to the basement for cabling. Those were on the front wall behind the speakers. Since the subs were not in that area and I did not want to run cables from behind the main speakers across the room to the locations of the subs, I decided to run wires from the equipment rack to the right wall behind the couch.

I cut an opening the size of an old low voltage box and ran the speaker wires through holes cut in the floor sill to the equipment rack. For each sub, I ran the appropriate length of Belden 5000UP in-wall speaker cable to the amplifier. When running wire in the wall be sure to run the appropriate type of wire that is CL2 or CL3 rated for fire protection. The wire that fed out from the wall was connected to a dual Speakon wall plate.

Photo 16 shows the wires connected to the lugs on the back of the Speakon connectors on the wall plate. **Photo 17** shows the wall plate mounted to the low voltage box. The connectors were labeled for the left and right subs.

The amplifier I used to power the subs was a Crown Studio Reference that is rated at 1,200W per channel into the 4Ω load of each enclosure. It certainly was far more power than I needed but it was available and not being used so I pressed it into service. Nothing like having headroom to

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Photo 16: I connected the wires to the Speakon plate.

Photo 17: Finally, I mounted the wall plate to a low-voltage box.

Resources

Dr. Earl Geddes, "Why Multiple Subs," GedLee, LLC, www.gedlee.com/Papers/multiple%20subs.pdf

T. Perazella, "Acoustic Room Treatments: Taming the Gorilla," *audioXpress*, August 2021.

T. Welti, "Subwoofers: Optimum Number and Locations," White Paper, Harman International Industries, Inc., www.harman.com/documents/multsubs_0.pdf

Sources

Dayton RSS265HF-8 10" drivers
Parts Express
www.parts-express.com/Dayton-Audio-RSS265HF-8-10-Reference-HF-Subwoofer-8-Ohm-295-442

Acousta-Stuf Polyfill 5lb bag cabinet stuffing
Parts Express | www.parts-express.com/Acousta-Stuf-Polyfill-5-lb.-Bag-260-330

spare. Depending on room size, a stereo amp in the 200W to 400W per channel rating should be sufficient.

After much thinking on how to get the appropriate signal to feed the subs I decided to take the low-frequency signal from the low output of the DEQX Express II processor that feeds the woofers in my main bass cabinets and use a splitter cable to send the signal to a Behringer DEQ2496 equalizer and from there to a Behringer DCX2496 DSP processor, both of which I already had. The connection from the DEQ2496 to the DCX2496 was via an AES/EBU cable. The audio output from that chain was then fed to the amplifier.

Integrating the Subs

The integration process consisted of both measurements and listening. It began by setting time delays and crossover points for the subs. To reiterate, the goal was to eliminate the suck out around 40Hz that occurred at the prime seating, not to get maximally flat bass response in the whole room. Given the placement of the subs, two different time delays were used.

The left channel was 3ms and the right channel 0.8ms. For the crossovers, both channels were Linkwitz-Riley with slopes of 48dB/octave. The turnover point for the left channel was 63Hz and the right channel 56Hz.

In addition, parametric EQ was added to both channels to further smooth the response. Final settings were for the left channel, 31.3Hz, ¼ octave, -6.5dB; 45.3Hz, ¼ octave, +4dB; 50.8Hz, 1/7 octave, +5dB; 55.7Hz, 1/9 octave, -6dB. For the right channel, 37.7Hz, 1/3 octave, +1dB; 44.8Hz, 1/10 octave, +2.5dB; 49.1Hz, 1/8 octave, +4.5dB.

The four 10" drivers were capable of substantial low bass, but that area was already well handled by the main woofers. To prevent overdriving the subs at very low frequencies and high drive levels, Dynamic EQ was employed in the DEQ2496. The turnover point was 34.8Hz with a gain of -10dB, a threshold level of -10dB, a ratio of 1:3.0, an attack time of 20.1ms and a release time of 304.2ms. It is unlikely that the Dynamic EQ will ever be needed but it is nice to have. The measured results at the prime listening position with all the adjustments are shown in **Figure 5**.

What Does It Sound Like?

Before integrating the subs into the room, the two major shortcomings were a lack of impact on some kick bass notes and an unevenness in intensity as string bass notes went up and down.

"Aeris" by Eric Godoy is a good test of both a plucked and bowed string bass with a full range of notes. With the subs, the levels were very consistent, even to the lowest notes.

The drum kit starting around 7 minutes into "Hey Mr. Mumbles" on *Live at the Village Gate* by Clark Terry is very dynamic and a good test of how well the bass reproduction can be. The kick bass sounded strong and tight.

The Dorian recording of Modest Mussorgsky's *Pictures at an Exhibition*, Track #2, "Gnomus" has great low notes that have a tight, rumbling sound that was now very evident.

Similar improvements were noted in other classical and pop recordings. As a result, I can say that the project, although quite involved, was certainly worth the effort.

Many will say that deep bass does not convey much of the musical information, but if you have ever heard deep bass at a live performance, you will dispute that notion. Clean, tight, low bass forms the foundation of most genres of music. If it is not present, a significant part of the emotional impact of the music is missing.

The low bass needs to be uniform up and down the scale to prevent either boominess or thinness in the music. Once there, it is very satisfying and helps convey the emotional message the performers are attempting to convey. As the old saying goes, "Try it, you'll like it!"

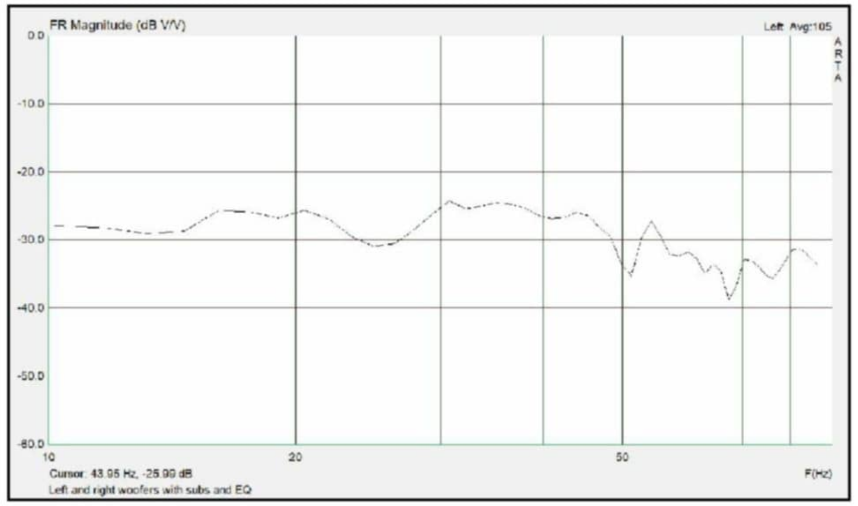
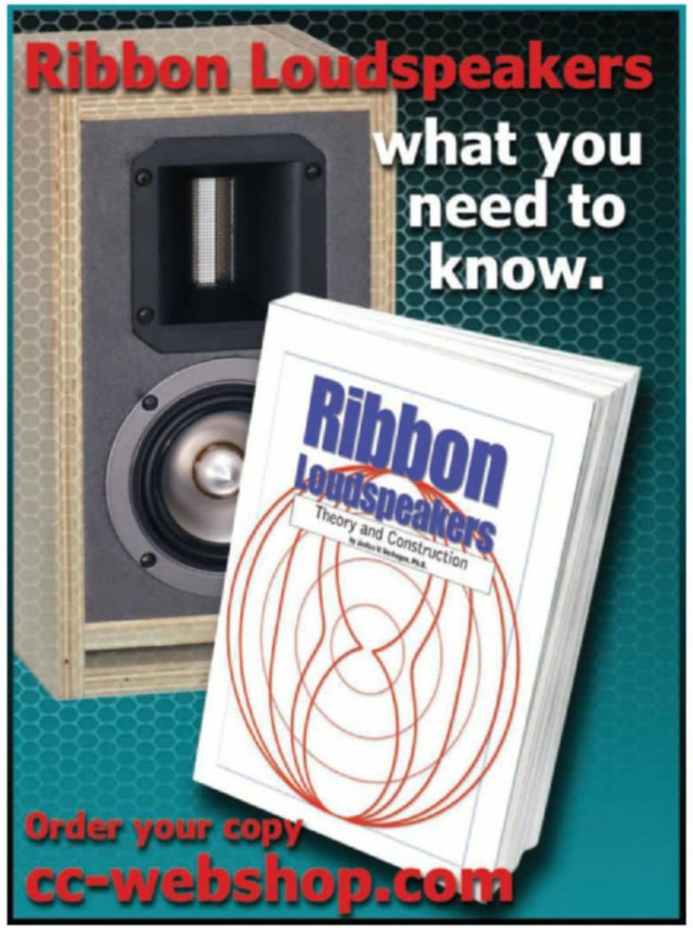


Figure 5: Here are the final measurements at the listening position.



DC-Coupled Microphone Preamplifier

This article explores avenues to improve microphone preamplifier circuits powering condenser microphones (**Photo 1**), and the different circuit options to decouple the microphone input from the preamplifier itself. The author proposes a high-quality design featuring a DC-coupled input with a floating supply.

By
Michel Nieuwenhuizen

Photo 1: Recorded organ music using the microphones as described in a previous audioXpress article series, along with the new microphone preamplifiers described here.

Most microphone preamplifiers have a provision for powering condenser microphones with 48V phantom powering via two 6.8kΩ resistors or a single 3.3kΩ resistor (**Figure 1** and **Figure 2**). A connected condenser microphone will draw a DC current from both input terminals between approximately 1mA and 12mA, while a dynamic microphone will not draw any current. As a result, the DC voltage between the inputs and ground can be anything between 0 and 48V (or 0 volts when the phantom power is switched off). It is not easy to design a preamplifier that can handle this wide range of DC input voltages, which is why there is normally a DC decoupling between the microphone input and the preamplifier itself, either via a transformer or (large) electrolytic capacitors (elcaps).

But top quality input transformers are expensive and elcaps are cheap but are prone to a relevant level of distortion. In the excellent articles by Cyril Bateman (see Resources), it is demonstrated that they distort quite a lot, especially when there is a large DC voltage across them. Expensive special elcaps are only slightly better. Bateman's solution is to use a series connection of two bipolar capacitors as the best option. However, even this option still shows some distortion and bipolar 100µF 50V capacitors are not easy to obtain.

Elcaps at the input also have a problem when the DC voltage at an input terminal suddenly changes (e.g., a short circuit in a cable). This causes a sudden discharge of the energy stored in the capacitor into the surrounding electronics, possibly destroying or degrading them. That is why you find large (1A) protection diodes at most preamplifier inputs.

Both problems could be solved when DC coupling of the preamplifier would be possible. Since we cannot (and do not want to) change the microphones [1], this requires a preamplifier that can function at all DC input levels between 0 and 48V. I will show in this article that such an amplifier is possible at relatively low cost.

But first I would like to draw attention to yet another capacitor problem: Any DC offset inside the preamplifier itself will be amplified with the same gain as the audio signal (up to 1000 times). This could result in an unacceptable high DC offset at the output of the amplifier. Normally, this is solved by adding a large capacitor Cg in series with the gain setting resistor Rg (Figure 1 and Figure 2). Cg must be very large (at least 1000µF), since Rg can be very low at high gain settings. As I will show, a DC coupled preamplifier does not need this solution since it will remove the DC offset in a different way.

A DC-Coupled Preamplifier

So the idea is to have a DC coupled input (**Figure 3**). For the amplifier to work correctly its supply should adapt itself to ±12V with respect to the average DC at the input (i.e., 0…48V). This results in optimal working conditions for the amplifier at any DC input level. In Rudolf Ondrejka's "A Novel Topology for a DC-Coupled Phantom-Powered Microphone," such an approach was given, but only in a rudimentary form ignoring a number of practical problems.

The solution presented here solves these problems resulting in a complete functional circuit, ready for practical use. However the DC at the output is now at the same DC level as the inputs so do we still have the same problem? No! This is because now we can design the circuit such that we can use a much smaller capacitor value (e.g., 2.2µF) for which there are very good low distortion options. The easiest and cheapest option is a high voltage (400V…1000V) polyester capacitor, but you might use any esoteric capacitor you like. We will now explain the circuit in detail.

Floating DC Supply

For the floating supply voltages we take the average DC of both preamplifier inputs with the resistors R1 (**Figure 4**). This average still contains all of the common mode AC from the input, which must be filtered out. A third-order 3Hz low-pass filter gives at least 50dB suppression at 50Hz/60Hz [2]. After the filter we have a stable DC voltage Va that we use as "ground" reference for the gain part and as centre voltage for the floating supply. This supply has as its own supply voltages of –15V and +63V and gives floating supply voltages of ±12V with respect to Va between 0V and 48V. The complete circuit is shown in **Figure 5**.

The output of the low-pass filter is the reference voltage Va. Transistors T1 and T2 are current sources so the voltages across the 2.7kΩ resistors is constant (approximately 13V), independent of Va. So:

$V{+}{+} = Va + 12V$ and $V{-}{-} = Va - 12V$ ($Va = 0…48V$).

The two red LEDs in Figure 5 form a simple overcurrent protection. I am not sure if they are

Figure 1: Transformer-coupled input

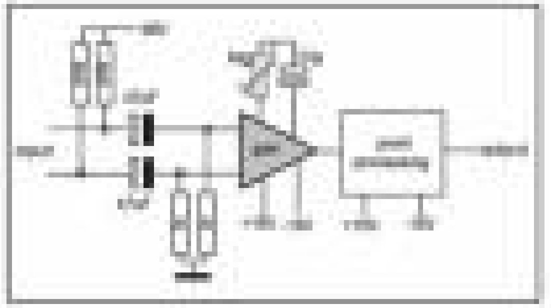


Figure 2: Capacitor-coupled input

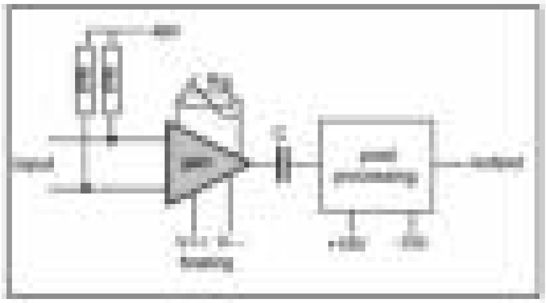


Figure 3: DC-coupled input

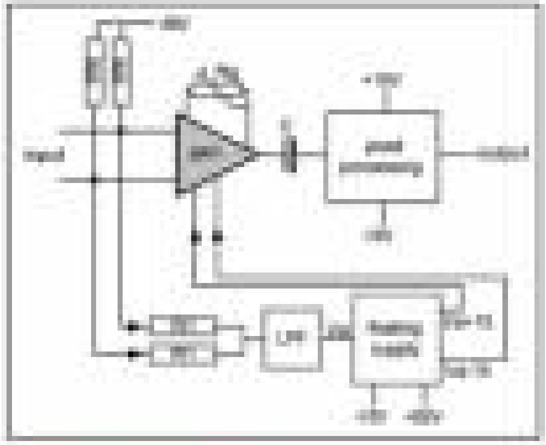


Figure 4: DC input with floating supply

About the Author

Michel Nieuwenhuizen studied electronics at the Delft [illegible] Technical University in The Netherlands. He then worked for 27 years with Philips Electronics in audio and video processing, resulting in a number of patents on video image [illegible]. After his retirement, he concentrated on audio projects such as a software package for automation of local radio stations called [illegible] and microphone preamplifiers.

really necessary, but some extra caution can't hurt.

Balancing

Now the preamplifier has an optimal power supply, but we also have to remove any differential DC offset between the inputs since that would be amplified with the gain of the preamplifier. Causes include:

- Input offset in the electronics of the gain part of the preamplifier
- Spread in the 6.8kΩ resistors of the phantom supply
- DC asymmetry between the outputs of the attached microphone

To compensate for this, we use an integrator that measures the output DC offset of the preamplifier, integrates it and (symmetrically) adds and subtracts a small current via resistors R2 to the inputs. In the specification "THAT 1580/5171 Digitally Controlled Microphone Preamplifier," a similar

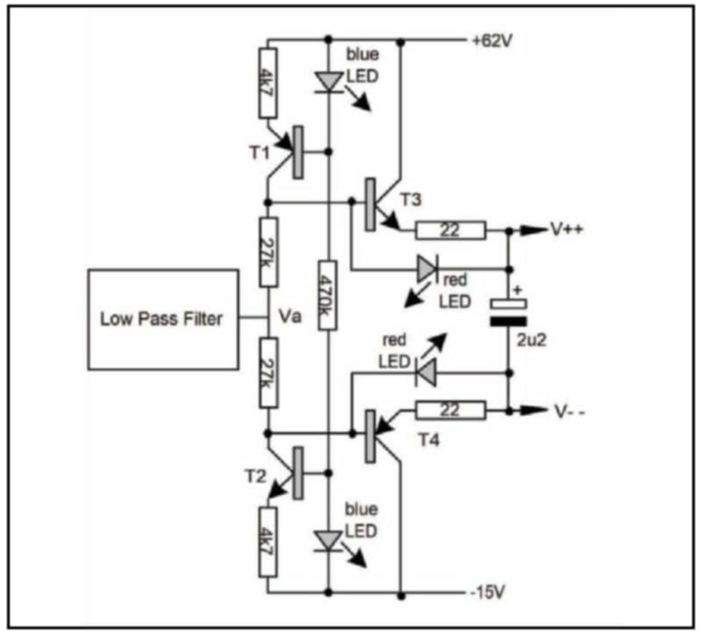


Figure 5: Circuit of the floating supply

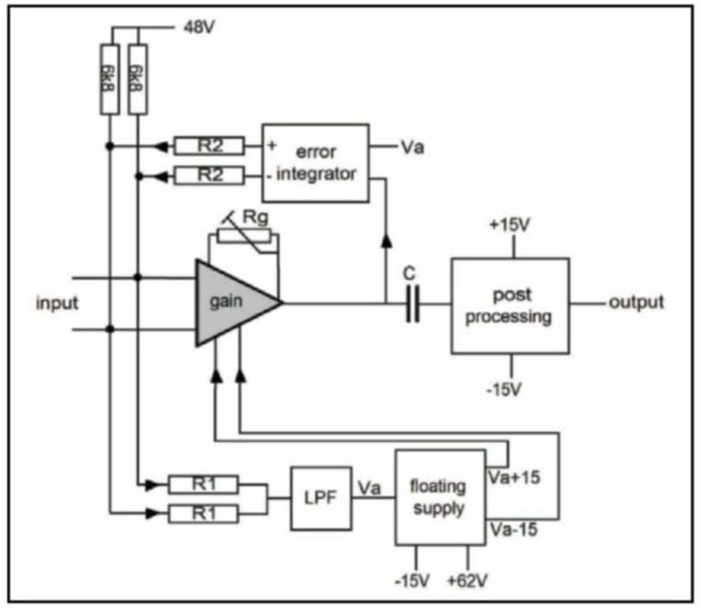


Figure 6: DC-coupled amplifier with floating supply and offset compensation

Author's Notes

[1] The Dutch firm Rens Heijnis, for example, adapts existing microphones by adding a fourth wire (or sometimes a fifth or sixth wire) between the preamplifier and the microphone, thus removing the phantom power on the audio wires. Strangely (as far as I can find from the Internet pages, because he supplies minimal and confusing technincal info), he still uses elcaps at both the microphone output and the preamplifier input, thus taking away part of the potential benefits of this (expensive) approach. I think that there are very good reasons not to go along this path, because apart from having to adapt all your microphones (which can then no longer be used in any other setup and of course making any guarantee void), you need expensive, non-standard cables. The performance benefit claims are not substantiated by measurements and when contacted, requests for further explanation are not answered.

[2] Here I have to explain that many modern microphones no longer have symmetrical outputs. Even top microphones from Neumann, Sennheiser, and others put the audio output on only one output pin while the other pin is made to have approximately the same output impedance but carries no audio. I call this semi symmetrical. This means that the audio signal is not only in the differential signal, but half of it is a common mode signal. The microphone preamplifier must deal with this and might behave differently. In any case, it means that preamplifiers should be tested and measured for both truly symmetrical and semi symmetrical conditions.

It is remarkable that almost all legendary microphones such as Schoeps, DPA, and others do not follow this trend and are truly symmetrical. To give an example: The older Neumann KM84 uses an output transformer and hence is truly symmetrical while its successor KM184 uses a semi-symmetrical output. The KM84 is much higher regarded than its replacement.

In our case, the suppression of the common mode signal should be extremely good, not only to filter out unwanted hum but also because it might or might not carry part of the audio signal.

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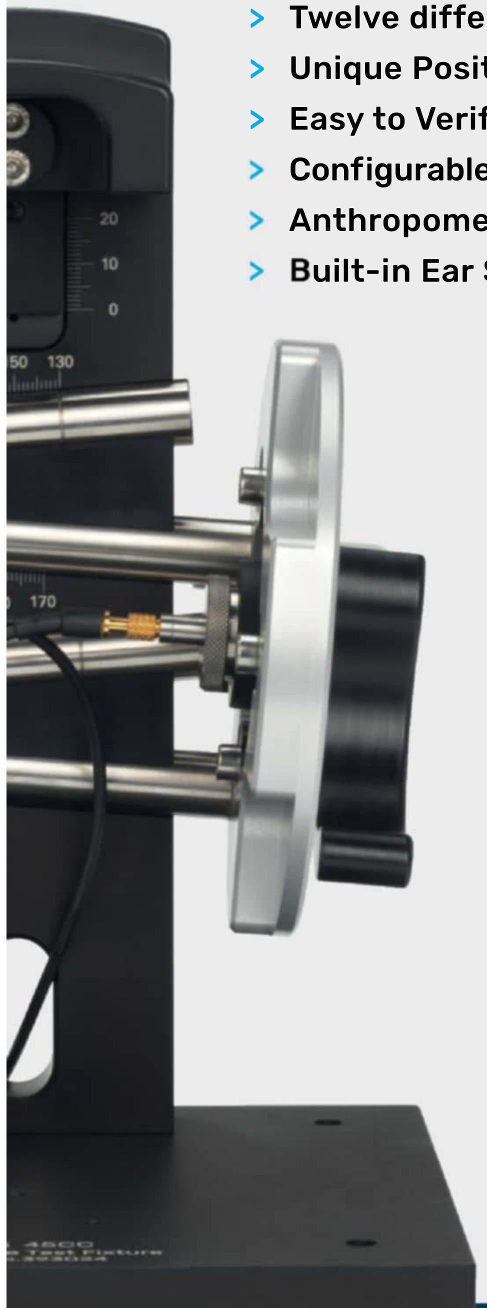


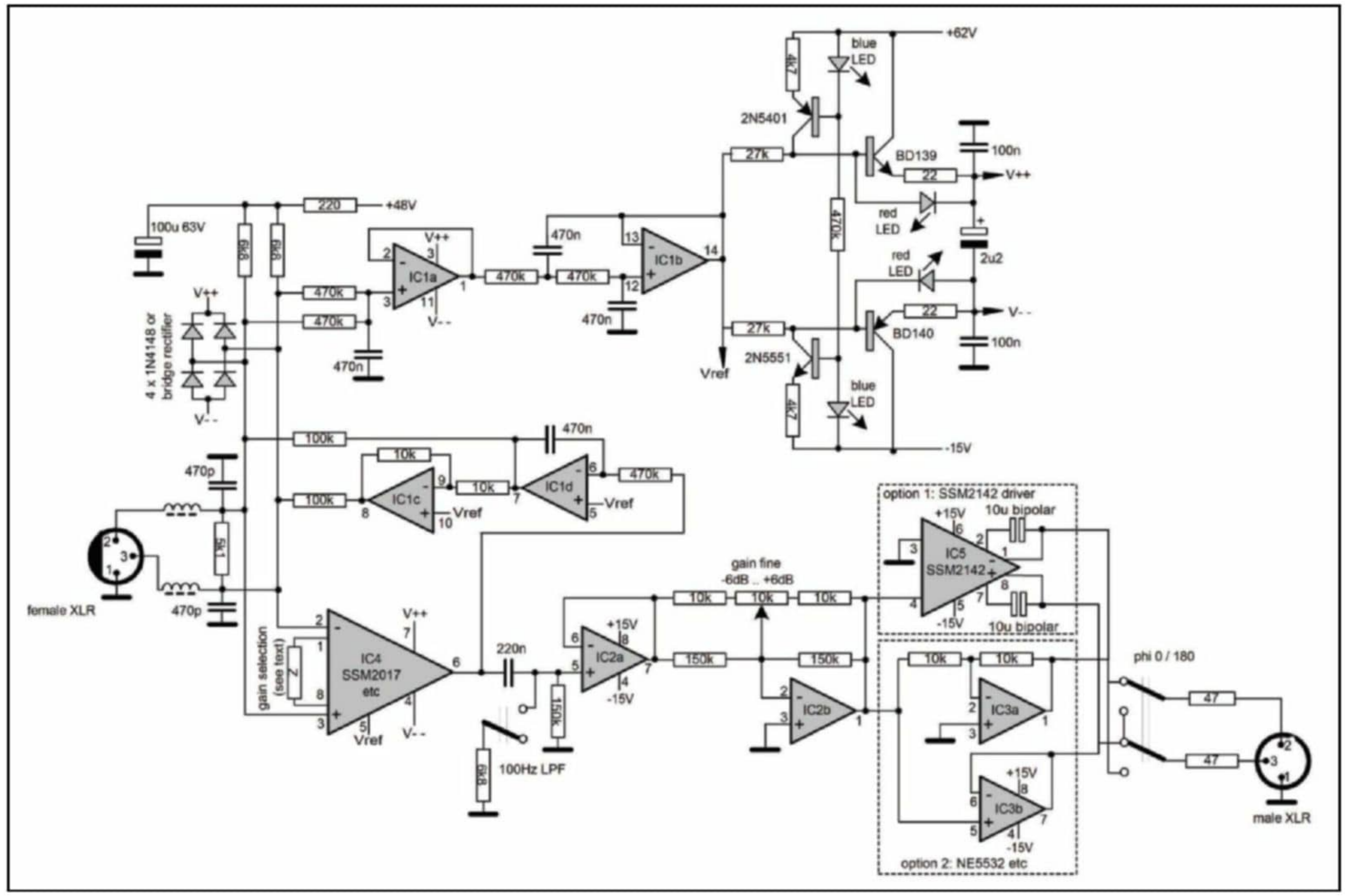


Figure 7: The complete circuit

error integrator is mentioned for compensation of the input offset of the preamplifier only, but I use it here to remove all input offsets. By adding/subtracting the compensation symmetrically, any coupling between the balancing and the floating power supply is avoided (**Figure 6**).

The Circuit

The complete circuit is shown in **Figure 7**. There is a simple RF filter at the input, similar to what is found on other preamplifiers. I also added diodes to make sure that the input DC does not go above or below the supply of the SSM2017. Since there are no big capacitors that can short-circuit, we do not need power diodes for protection. Standard 1N4148 diodes or a low-power bridge rectifier are sufficient. The 6.8kΩ phantom power resistors should be matched as closely as possible.

The gain block can be any normal microphone preamplifier with plus and minus supply voltages of 10..18V, with respect to the local ground. I use the single chip SSM2017 amplifier (now discontinued) because I had some lying around. There are a number of similar alternatives (THAT1510, THAT1512, INA163, and SSM2019) some with different pin-outs. With a price tag of $4 or less it is hard to beat these chips on value for money but you might use any integrated or discrete amplifier as long as it has similar power requirements.

The gain is set with variable resistor Z. For a good "feel" Z should have a negative logarithmic control characteristic. Since neg-log potmeters of good quality are hard to find, I prefer to use a switched resistor network. In **Figure 8** are two examples for use with the SSM2017. Gain increases with 4dB per step giving a range of approximately 20 .. 60dB.

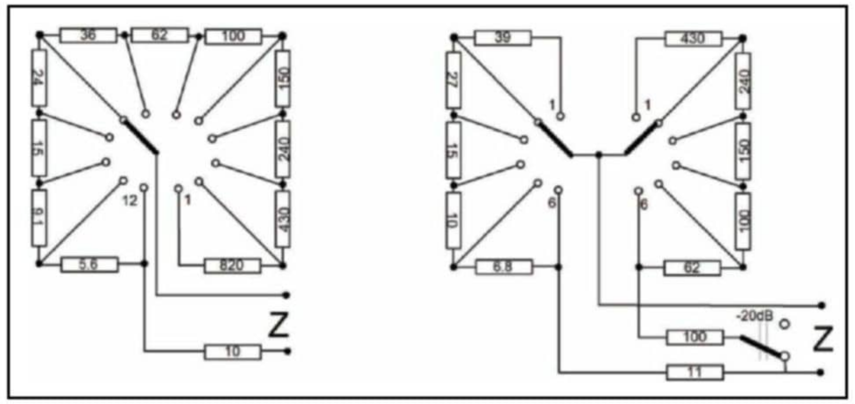


Figure 8: Gain defining Z (left with a 12-position switch, right with dual 6-position switch)

After the gain stage, we have the only capacitor in the signal chain. The buffer stage after this capacitor should be a FET-based op-amp to avoid offsets. The OPA2134 is one of the most musical op-amps available and not expensive.

The remainder of the post-processing is standard and may be adapted to your liking. I added a first-order 100Hz low-pass filter switch because it is almost free and does not add distortion. Since I use a stepped gain setting in the preamplifier, a "fine gain" control of ±6dB is added. I give two options for the output.

The SSM2142 is especially useful when long cables are connected to the output but requires two bipolar elcaps for low DC offset (or you have to accept a possible DC output offset of maximum 250mV). For short connections the NE5532, the OPA2134 or a similar op-amp might be better and less expensive. For very low offset voltages, the OPA270 is also an excellent choice. **Photo 2** shows the PCB's top (a) and bottom (b).

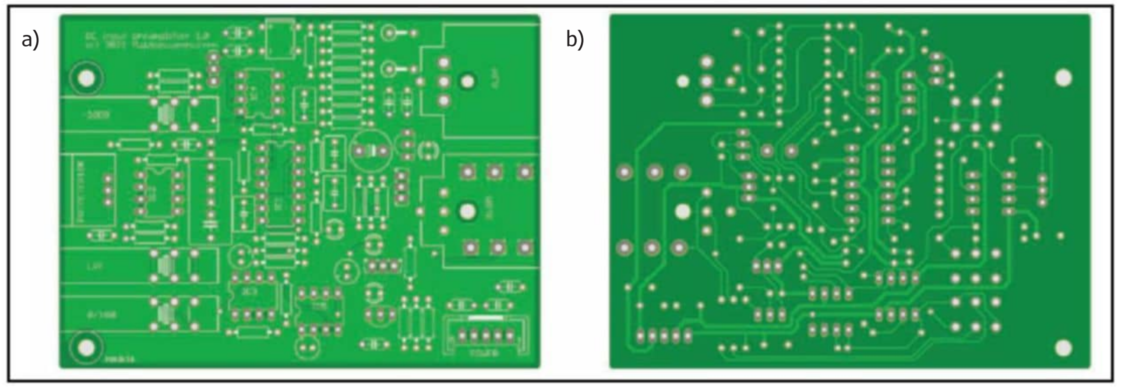


Photo 2: Here is the PCB's top (a) and bottom (b) layout. Not shown to scale.

Supply

The supply for this circuit is slightly more complicated than normal because of the extra 62V. **Figure 9** shows an example suitable for a few preamplifier channels. If you need more channels of preamplifier, you might prefer double-phase rectifying with three or four secondary transformer windings. The supply stabilizing ICs should have sufficient passive cooling of course.

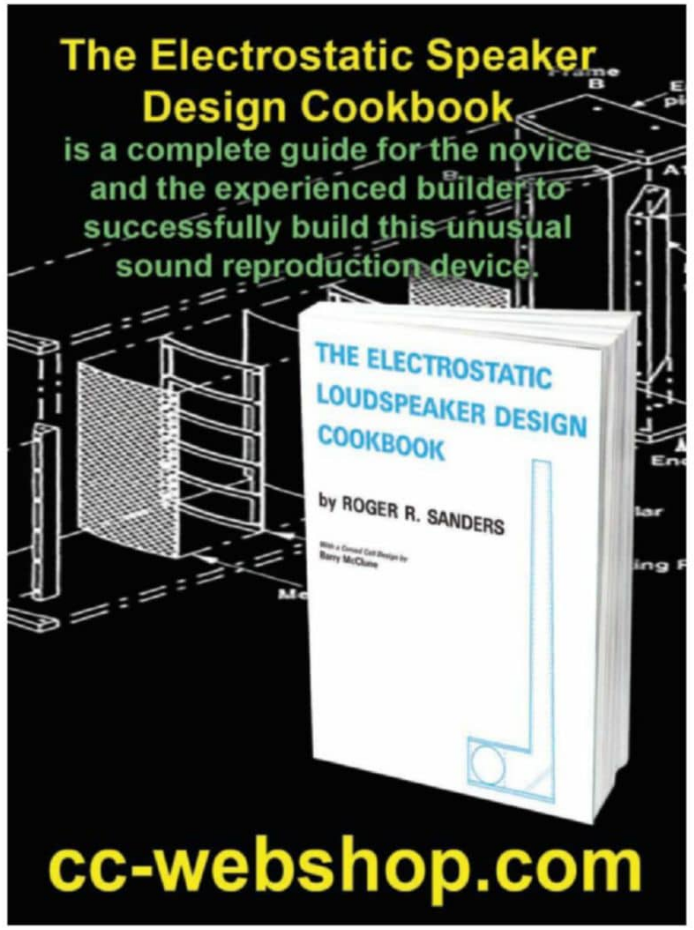
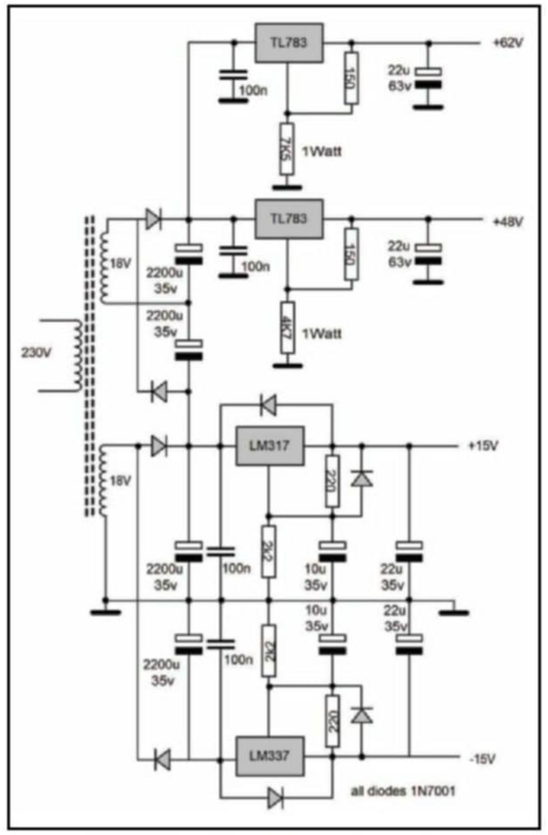


Figure 9: Here is an example of a possible power supply.



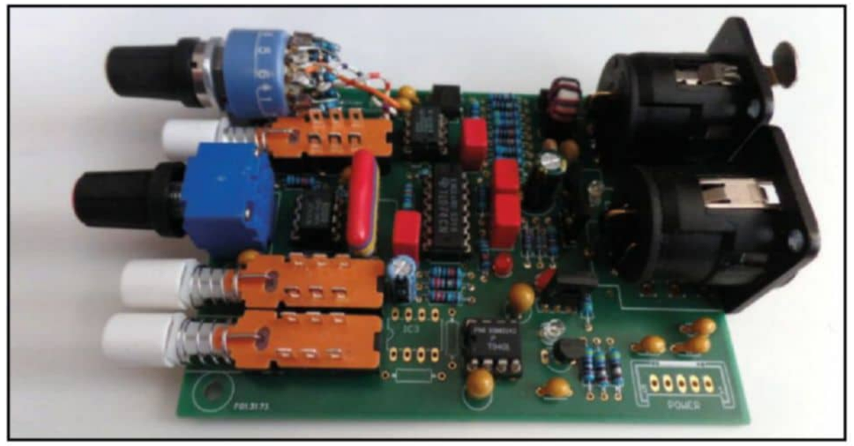


Photo 3: The completed double-sided PCB

Tests

A double-sided PCB was developed and tested (**Photo 3**). It measures roughly 100mm × 75mm and contains all the components apart from the stepped gain setting, which is connected via three wires to the board. Distortion measurements are of little use, because they depend almost completely on the specific preamplifier chip used. Listening tests reveal a nice, clean sound.

Microphone Safety

Since this amplifier has a DC-coupled input, we should carefully check any possible fault condition that might cause a DC current through a connected microphone. Dynamic microphones are especially vulnerable.

First, I tested the maximum current through a microphone during switch on of the power supply or the plug-in of a microphone. The maximum current was less than 10µA and lasted less than 5 seconds. Once stable the current dropped to less than 1µA. This is completely safe for all microphones.

Next I checked what might happen if there is a fault in the cable between the microphone and the amplifier. The only possible problem is when there is a short circuit between one of the signal wires and ground. In that case, there will be a maximum DC current of about 7mA, which is identical to the fault current in any other amplifier with phantom power. Of course if you want to be 100% safe, you might switch off the phantom power when using dynamic microphones.

I also tested the circuit with a number of professional microphones from Neumann, Sennheiser, and AKG. No issues were found.

For my recording of classical music, I built a rack of 3× five amplifiers with a Motu 16A interface as AD converter. **Photo 4** shows a strip of five amplifiers in the 19" module. **Photo 5** shows my 15-channel recording rack. The Motu has DC inputs, so now my whole recording chain is free from electrolytic capacitors.

Conclusion

This new way of connecting a phantom-powered microphone to a preamplifier provides the benefits of better sound quality and better protection against cable defects. The extra components are easily obtained and relatively inexpensive. Audio performance is good and primarily determined by the gain stage of which there is ample choice.

Resources

C. Bateman, "Capacitor Sounds 6: 10 to 100µF capacitors and 100Hz measurements," (updated & extended March 2003), *Linear Audio*, April 2017 (Original version published in *Electronics World*, January 2003), https://linearaudio.nl/cyril-batemans-capacitor-sound-articles

C. Bateman, "Capacitor Sounds 4 - 100nF to 1µF," (updated & extended March 2003), Linear Audio, April 2017 (Original version published in Electronics World, November 2002), https://linearaudio.nl/cyril-batemans-capacitor-sound-articles

R. Chalupa, "A Novel Topology for a DC-Coupled Phantom-Powered Microphone," *Journal of the Audio Engineering Society* (JAES), October 1989.

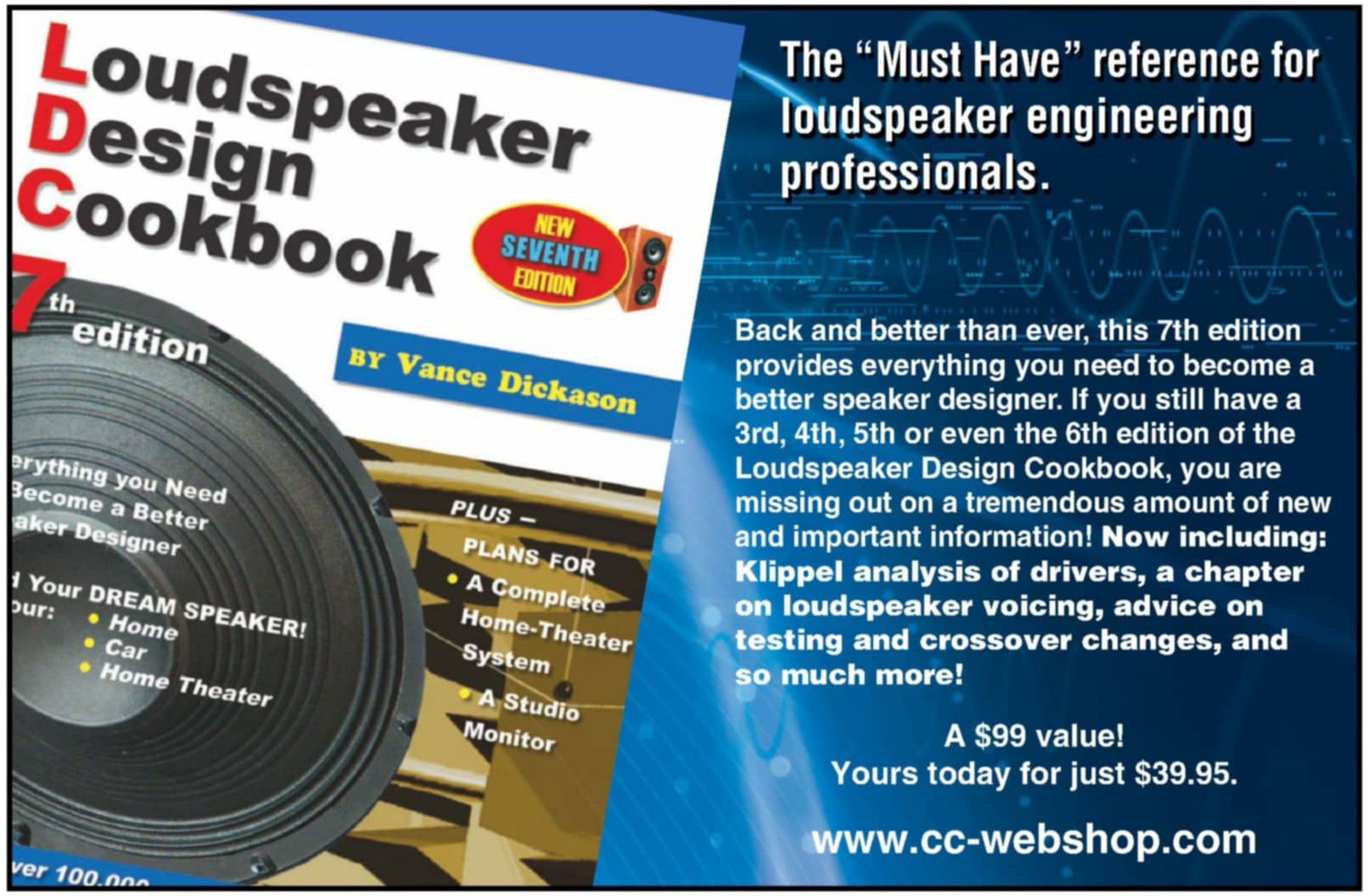
M. Nieuwenhuizen, "Ultra-Low Distortion Microphone Buffer Amplifier (Part 1): Construction," *audioXpress*, October 2020.

M. Nieuwenhuizen, "Ultra-Low Distortion Microphone Buffer Amplifier (Part 2): Simulations and Measurements," *audioXpress*, November 2020.

"THAT 1580/5171 Digitally Controlled Microphone Preamplifier," THAT Corp., 2020 www.thatcorp.com/DigMicPre.shtml

Photo 4: This is a single strip of five amplifiers that I've used in the 19" module.

Photo 5: For my recording of classical music, I built a rack of 3× five amplifiers with a Motu 16A interface as AD converter.



"Pass-Able" Fully Discrete Line Stage

By
Aren van Waarde

Photo 1: The preamplifier in its recycled enclosure. (Spurious holes in the front panel were hidden behind a piece of cardboard. If I have time, I may prepare a novel aluminum front—but I tend to be lazy!).

To play music via a commercial power amplifier, the author built a line-level preamplifier with an "old-fashioned" gain stage using only discrete transistors, inspired by a Nelson Pass article. Combined with a quiet power supply and a small PCB-type power transformer, the results surprised positively.

In order to play music via a commercial power amplifier (which seemed broken, but could be repaired), I needed a line-level preamplifier. Since our country was in lockdown because of COVID-19 restrictions, I decided to build a preamp using only parts that were present in my junk box (**Photo 1**).

The power amp in question is a heavy (12kg) solid-state model without volume or level controls, very solidly built using high-quality parts and conservatively rated as 2× 110W RMS and more than 2× 150W peak in 8Ω. After cleaning its innards of the spider webs and fixing a few screws on the power supply capacitors, it functioned again and seemed to operate flawlessly. Thus, I wondered how it would sound when it was coupled to a suitable preamp.

Gain Stage

After several SPICE simulations [1], I opted for a circuit layout proposed by Nelson Pass in his lucid and entertaining article "DIY Op Amps" [2]. The actual op-amp consists of only four transistors: a differential amplifier (Q3,Q4) is followed by a second gain stage in common collector mode (Q1) and a voltage follower (Q2), all running in Class A (**Figure 1**).

In order to improve the linearity and to increase the power supply rejection ratio (PSRR) of this minimalist op-amp, I added two constant current sources (Q6,D2,D4 and Q5,D1,D3)—a modification that was suggested in Nelson's article. The first current source (Q6,D2,D4) is set to approximately 2mA and feeds the differential amplifier, whereas the second CCS (Q5, D1, D3) keeps the current through Q1 constant at about 5mA. For the voltage follower (output transistor Q2), I did not use a constant current source (CCS), instead I used an emitter resistor of 3kΩ 2W (R10). This resistor dissipates 0.3W continuously, since about 10mA runs through Q2. During normal use, Q2 becomes only slightly warm. The use of a heatsink for this transistor is therefore not strictly necessary.

The only additions that I have made to the basic circuit proposed in Nelson's article are: the input capacitor C3 (and volume control VR1); C5 between circuit input and ground, which forms a low-pass filter with R3; C6, the capacitor in parallel to the negative feedback resistor R4; and R5, C4, and R8 at the op-amp output. The values of C5, C6, and R5 were based on SPICE simulations.

If the power amplifier that receives its signal from the op-amp has an input capacitor, C4 may be removed and be replaced with a wire link, but I do not recommend this. Without C4, some DC offset will be present at the output of the gain stage. I measured 107mV at the left channel and 108mV at the right channel of my prototype, thus, the DC offset is considerable.

VR1 is the volume potentiometer (10kΩ log) at the input. As output (C4) caps I used cylindrical MKC (polycarbonate) types that were manufactured in the German Democratic Republic. These vintage items are sonically excellent. They have very little coloration. C6 was added to ensure stability of the feedback loop, and R5 was included to isolate the gain stage from the load. R8 was added to discharge the output capacitor, in order to avoid annoying pops from the loudspeakers. I used what was available in my junk box, thus Q2...Q6 are other transistors than were described in Nelson's article [3] Q3 and Q4 were "paired" (selected from a large batch of transistors for identical current gain), using the procedure that Nelson proposed. They were also thermally coupled (by supergluing the flat side of the two transistors together).

I designed and etched a PCB (**Figure 2**) [4] and after I had mounted all components, the op-amp seemed to work well. However, when I tested the PCB with a sine and square wave generator and monitored the output with an oscilloscope, the initial results were disappointing.

The oscilloscope images indicated a (beautifully sinusoidal!) RF oscillation of the circuit at a frequency of about 7MHz, which would definitely spoil its sound... After some head scratching, I noticed that I had forgotten to install the power supply bypass capacitors C1 and C7 on the PCB. Thus, I soldered four ceramic multilayer capacitors of 100nF between the positive and negative supply rails and board ground (two for each stereo channel, at the solder side of the board). After this addition, the gain stage was completely stable and fully quiet in terms of hum and noise.

Power Supply

Nelson Pass proposed power supply rails of ± 32V for his DIY op-amps. A suitable PSU could be

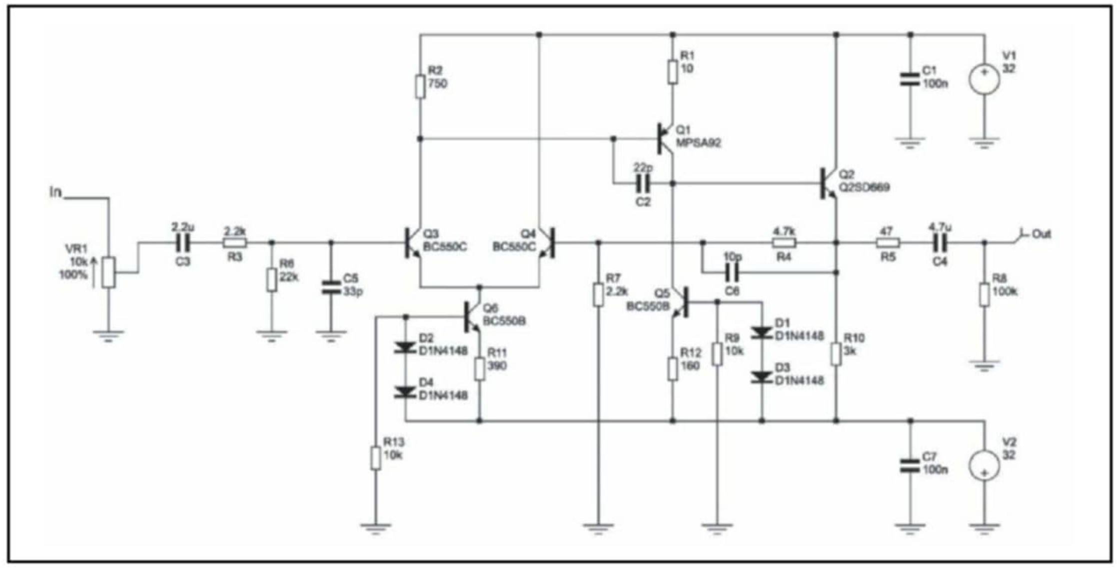


Figure 1: Schematic of the gain stage. Only a single channel is shown. (Editor's Note: At the request of *audioXpress*, Nelson Pass commented: "It's a perfectly fine design. As Mr. van Waarde, modestly has credit me with the circuit, I will follow his example and state that it's a classic circuit, not representing a lot of originality on my part...")

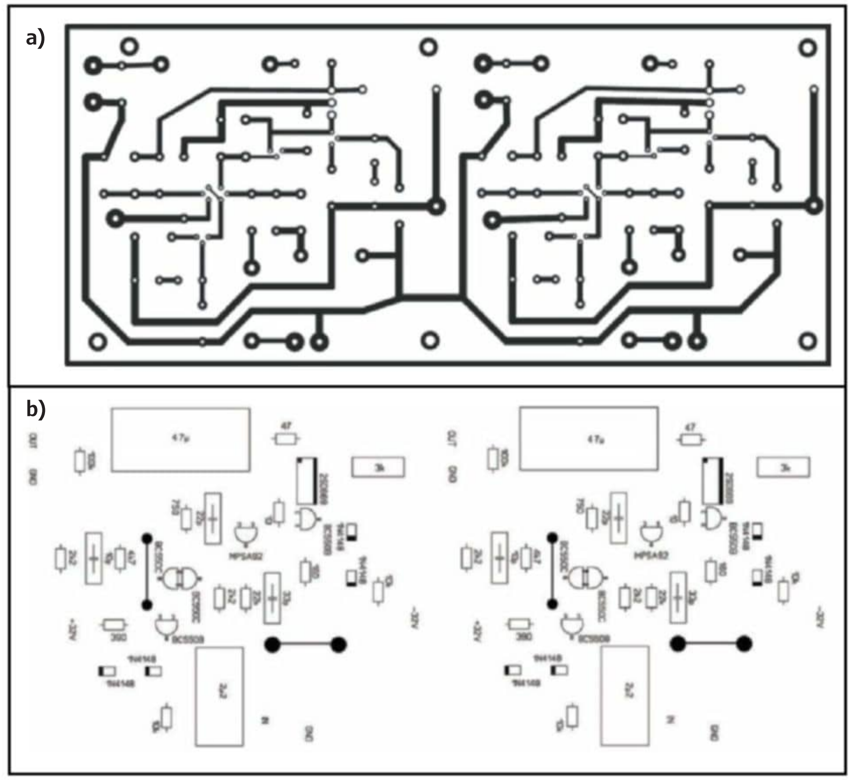


Figure 2: The amplifier board with the etching pattern is shown from the component side (a). The amplifier board with parts placement is shown from component side (b). Not shown are the capacitors C1 and C7, which (for each channel) were mounted below the board, between the pads marked +32 and -32V and the ground track. Note the four wire links (these should not be forgotten). The actual size of the board is 156mm × 70 mm.

About the Author

Aren van Waarde has a PhD degree in animal physiology from the University of Leiden. He worked as a postdoc at Yale University, as a fellow of the Royal Dutch Academy of Sciences in Leiden, and for the last 32 years as a biochemist at the University Medical Center Groningen in the field of medical imaging (positron emission tomography). In his spare time, Aren builds amplifiers, loudspeakers and receivers. When he isn't listening to music or writing a book, he may be on a hiking tour to take close-up pictures of wild flowers, mushrooms and lichens.

made by using a LM317-LM337 pair, but I considered the application of integrated circuits (ICs) to feed a discrete op-amp "a damnable heresy." Thus, I decided to build a ± 32V power supply using discrete transistors (**Figure 3**). This circuit was applied by the (now defunct) Norwegian company Tandberg in its TCA3002 "Control Amplifier." A possible PCB layout that I designed is shown in **Figure 4**.

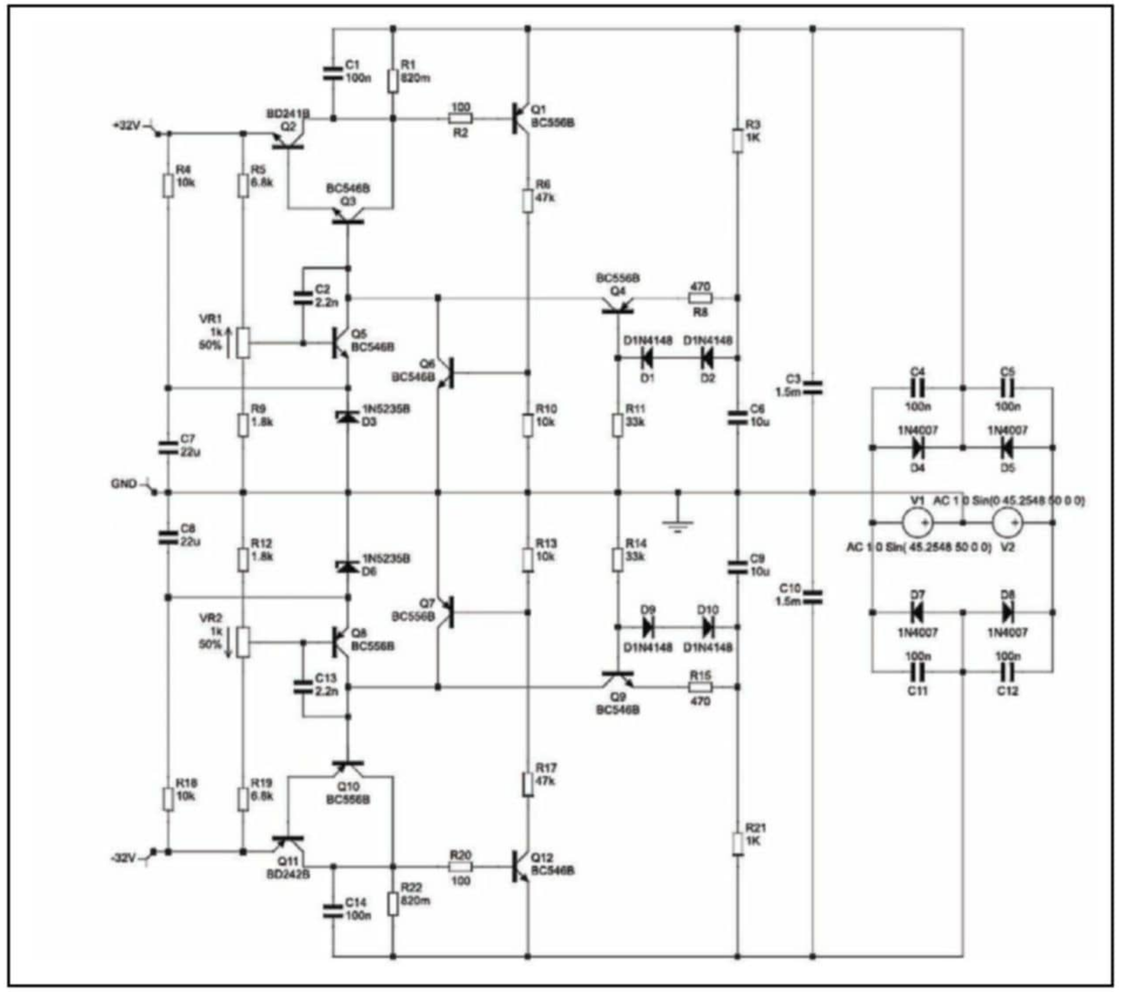


Figure 3: Schematic of the power supply. This circuit feeds both preamp channels.

Author's Notes

[1] For computer simulations of electronic circuits, I use the program SIMetrix (SIMetrix Technologies Ltd., Thatcham, Berkshire, UK). This software package is very user-friendly and well-documented. A demo version (with limited but generous circuit size) can be downloaded from the company website https://www.simetrix.co.uk and be used indefinitely.

[2] Nelson Pass, "DIY Op Amps" article can be found at www.passdiy.com.

[3] Nelson Pass used MPSA18 transistors for the differential amplifier, MPSA92 for Q1, and MPSA18 in his proposed constant current sources. He did not specify any transistor type for Q2. MPSA18 is not a common transistor in the Netherlands, thus, I replaced this by BC550C (for the differential amplifier) and BC550B (for the current sources). Since Q2 dissipates about 340mW continuously, I placed a medium power transistor with a TO-126 outline at this position (2SD669A, made by Hitachi, was present in my junk box).

[4] For PCB design, I use Sprint Layout (version 6.0). This German program is user-friendly, contains pinout data for very many standard components, and is not expensive. For information, you may contact Abacom Ingenieurgesellschaft (near the city of Bremen), www.electronic-software-shop.com.

[5] The only comparison that I made was the following: I compared the sound of the DIY op-amp with the sound of a passive preamp (which contained only a 10k log potentiometer of good quality). For reasons that I don't understand, I preferred the sound of the active preamp, which seemed more lively and dynamic. It's apparently not a good idea to connect the wiper of a potentiometer to a power amp via a long coaxial wire.

VR1 and VR2 are used to adjust the voltages at the power supply rails. I set the output voltage to ±30 rather than ±32V, since the small power transformer that was present in my junk box provides 30-0-30VAC and the PSU circuit needs some voltage drop between input and output for proper regulation.

The amplifier and PSU boards were mounted in a small metal cabinet (steel and aluminum, recycled from a discarded signal generator), whereas the power transformer (230V:30-0-30V, 8VA) was placed in a separate plastic enclosure that has once housed a modem for an 8-bit computer (**Photo 2**).

The input and output connectors of the preamp are insulated RCA jacks. A single wire connects the metal cabinet with the GND pin on the power supply board (**Photo 3**). The transformer and preamp enclosures are connected by a three-wire cord of 60" length. The transformer can thus be placed at considerable distance from the preamp, which may result in an improvement of the preamp's signal-to-noise ratio. **Photo 4** shows a top view of the power supply board, and volume controls for each channel.

Sound

I cannot make any competitive claims for the performance of this marriage of Californian and Norwegian technology [5], but after several listening tests I must say that I love its sound. It just sounds right (three-dimensional, detailed, lively, with considerable "foot tapping power"—you Naim it).

The XYL (aka my wife Maria, my most strict and influential critic) agrees with this positive evaluation. In our opinion, high-quality recordings

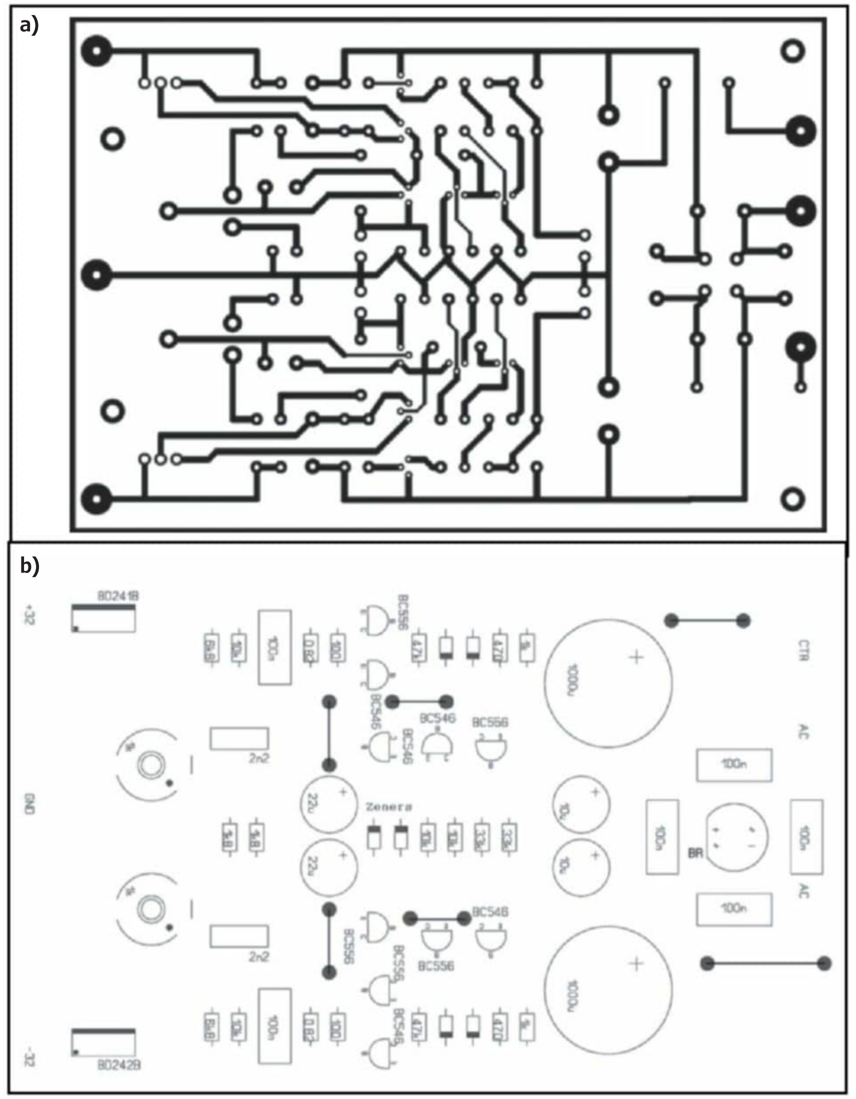


Figure 4: The power supply board with the etching pattern is shown from the solder side (a). The power supply board with parts placement is shown from the solder side). Note the six wire links (these should not be forgotten). The actual size of the board is 120mm × 82 mm.

Photo 2: This recycled modem cabinet houses the small 30-0-30V, 8VA PCB-type power transformer. The modem had metal front and back panels, which I replaced by novel ones made of black plastic.

Photo 3: Inside view of the preamp, with power supply board mounted at the bottom and amplifier board mounted vertically at the inside of the back panel. The big yellow caps marked "100V MKC, 4.7 µF" are the ancient German output capacitors. The thick white wire connects the metal cabinet to the GND pin on the power supply board.

of vocal music and speech should always be used for listening tests. Human voices should sound natural and sibilants should not be artificially emphasized. Another important amplifier test is the reproduction of the sound of harpsichords and violin strings. If a good recording of a harpsichord gives you the creeps, an amplifier problem may be present. Recordings of strings or massed strings can sound "hard" or "steely," but should sound sweet if the recording, the performing ensemble and the amplifier are good.

Sibilants of the human voice and the harmonics produced by harpsichords are poorly reproduced by many solid-state amplifiers. The DIY op-amp proposed by Nelson Pass (and my restored power amplifier) reproduce such sounds without any obvious artifacts. The preamp (and the restored power amp) are also completely quiet. Hum and noise are inaudible (even with an ear in front of the loudspeakers, at a distance of an inch).

In the 1960s and 1970s, amplifier manufacturers would have proudly advertised this project as a: "High-Definition Stereo Preamplifier. 24 silicon transistors, 16 silicon diodes, 2 Zener diodes". Those were the days. I was amazed by the discovery that such old-fashioned technology can sound so good.

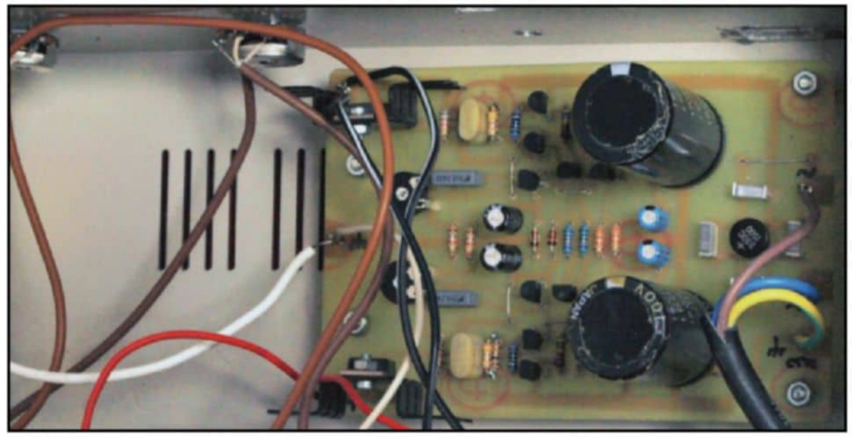


Photo 4: Top view of the power supply board, and volume controls for each channel. The output transistors of the power supply have small heatsinks. The input electrolytics of the power supply are recycled but measure still OK in terms of capacitance and ESR.

Parts List for the Amplifier PCB

Capacitors

C1, C7, C1a, C7a	100nF multilayer
C2, C2a	22pF ceramic
C3, C3a	2.2µF foil type
C4, C4a	4.7µF foil type (I used MKC)
C5, C5a	33pF styroflex
C6, C6a	10pF ceramic

Diodes

D1..D4, D1a..D4a	1N4148 (or 1N914)

Potentiometers

VR1, VR1a	10k log mono potentiometer

Resistors * With exception of R10, all resistors are 0.6W (I used mostly metal oxide 1%)

R1, R1a	10Ω
R2, R2a	750Ω
R3, R3a, R7, R7a	2.2kΩ
R4, R4a	4.7kΩ
R5, R5a	47Ω
R6, R6a	22kΩ
R8, R8a	100kΩ
R9, R9a, R13, R13a	10kΩ
R10, R10a	3kΩ, 2W (I used 2× 1.5kΩ 1W in series)
R11, R11a	390Ω
R12, R12a	160Ω

Transistors

Q1, Q1a	MPSA92
Q2, Q2a	2SD669
Q3, Q4, Q3a, Q4a	BC550C
Q5, Q6, Q5a, Q6a	BC550B

Parts List for the Power Supply PCB

Capacitors

C1, C14	100nF
C2, C13	2.2n
C3, C10	1500µF 63V radial (you may also use 1000µF or 2200µF)
C4, C5, C11, C12	100nF, 100V
C6, C9	10µF, 63V radial
C7, C8	22µF, 63V radial

Diodes

D1, D2, D9, D10	1N4148
D3, D6	1N5235B (any Zener diode 6.8V, 0.5W)
D4, D5, D7, D8	Small circular bridge rectifier (e.g, B50R DIO, in Figure 3 shown as 1N4007)

Potentiometers

VR1, VR2	1k trimpot, large horizontal type
V1,V2	30-0-30 VAC (supplied by outboard 8VA PCB-type transformer)

Resistors *All resistors are 0.6W (I used mostly metal oxide 1% types)

R1, R22	0.82Ω
R2, R20	100Ω
R3, R21	1kΩ
R4, R10, R13, R18	10kΩ
R5, R19	6.8kΩ
R6, R17	47kΩ
R8, R15	470Ω
R9, R12	1.8kΩ
R11, R14	33kΩ

Transistors

Q1, Q4, Q7, Q8, Q10	BC556B
Q2	BD241B [with small heatsink, **Photo 4**]
Q3, Q5, Q6, Q9, Q12	BC546B
Q11	BD242B [with small heatsink, **Photo 4**]

Building a Guitar-Controlled Synthesizer

VCA & VCF

By
Ethan Winer

Ethan Winer learned how to design electronic circuits by piecing together smaller building blocks using knowledge that he gained from various electronics magazines and from the application notes from semiconductor makers. For the past four issues our author has shared the modules he used to build a guitar-controlled synthesizer. In this final article in the series, he describes how to make two types of voltage controlled amplifiers plus the classic Moog low-pass filter.

As part of my ongoing guitar synthesizer project, this article presents and explains the final two key modules needed by every analog synthesizer: the Voltage Controlled Amplifier (VCA) and Voltage Controlled Filter (VFC). The control voltages that set the volume and filter frequency come mainly from the Attack Decay Sustain Release (ADSR) envelope generator shown in the June, 2022 issue of *audioXpress*. But the low-frequency oscillator (LFO) from that same issue is also used to provide a tremolo effect when driving the VCA, or to sweep the filter cyclically to add animation.

I'll show two different VCA circuits because one is much simpler, and works fine for one-off projects where precise unit-to-unit repeatability isn't needed. The second one is similar to the VCA in the original Moog synthesizers, using a topology similar to the Moog VCF. But the VCAs are simpler so let's start there.

VCA #1

In the 1970s, integrated circuits (IC) manufacturer National Semiconductor published a wonderful series of application books featuring its ICs and other products. National's sales reps would give these paperback books away for free to electronic engineers, and I still have the complete series and refer to them often. The two I use most are the *Linear Applications Handbook*, and the *Audio-Radio Handbook*. But others in the series discuss TTL and CMOS digital logic, and both junction and field-effect transistors (JFETs and FETs). The first VCA circuit shown in **Figure 1** is adapted from Linear Application Note AP129.

Although this circuit is simple, it has a gain range of 60dB—unity when the FET's gate is at -6V, to +60dB when the gate is at 0 volts and the FET's resistance is 30Ω. One downside is the ratio between control voltage and FET resistance isn't linear. Further, the control voltage range can vary between FETs, even from the same batch, and especially between FET types. So the range -6V to 0 volts used in this LTspice emulation might be -4V to -1V, or some other range, with the FETs you buy. However, it works perfectly well as the VCA in a volume compressor where the correct control voltage is self-seeking based on the current audio output level (**Table 1**).

The distortion of this circuit is too low to measure reliably in the spice simulation, so it's likely suitable for all but the most demanding hi-fi applications. Note the 60dB input attenuator that limits the signal level into U1 to 20mV to avoid clipping the output at maximum gain. If you don't need the full 60dB gain range, raise R2 to 4.7kΩ and lower R3 to 3.3kΩ. This limits the range to only 40dB, but improves the signal-to-noise ratio (SNR) ratio by 20dB.

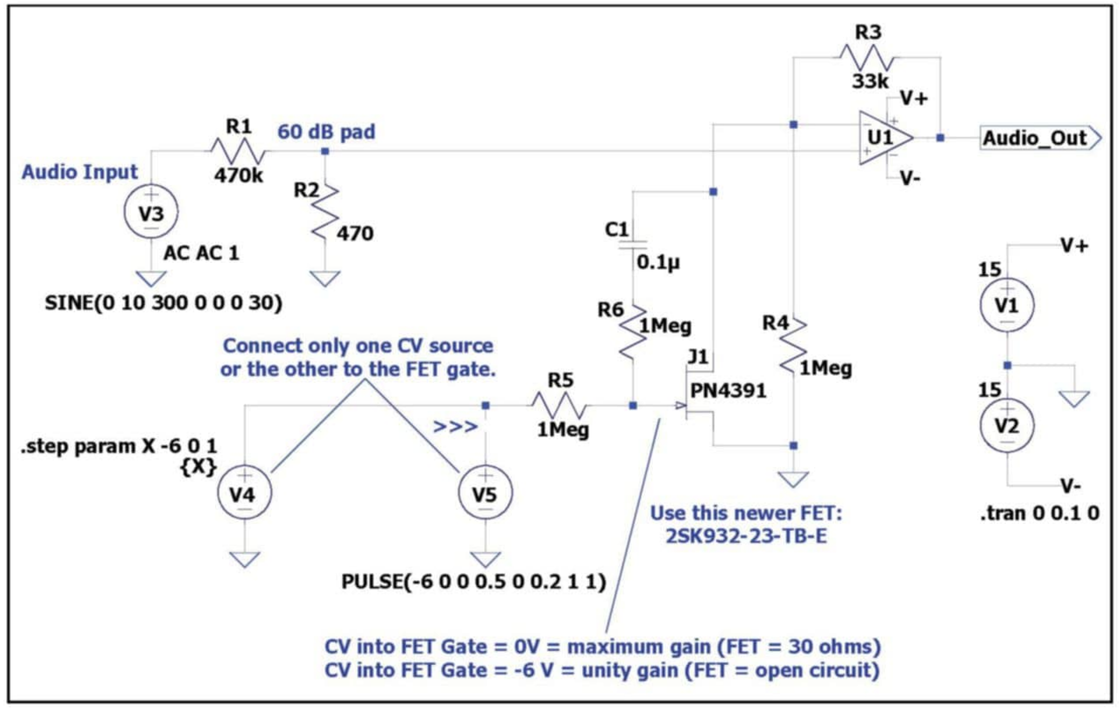


Figure 1: This VCA is extremely simple using a FET as a variable resistor, yet it allows a total gain range of 60dB.

One problem with LTspice is that it doesn't have all the latest semiconductor models in its component library. In this case the PN4391 FET is obsolete, so use the newer 2SK932-23-TB-E or almost any other small signal N-channel JFET. Likewise, noise in the op-amp can be a factor, so choose a current model such as the Texas Instruments OPA series.

I set up this circuit using two control voltage sources to provide different views into its operation. The file you can download is set to show the output for seven different control voltages at once when you probe the Audio_Output terminal. Each voltage is shown in a different color using the .step command. With .step, you specify a range of values for pretty much any circuit condition, and LTspice runs a series of simulations, one for each value in the range.

You simply assign a variable name (e.g., X or Y) to a component value, then tell .step to increment that variable. In this case, we're stepping the output voltage from source V4, but .step can vary resistor and capacitor values, and most other component attributes. If you right-click the .step command line, a dialog pops up letting you specify useful options such as linear or log steps; starting, ending, and step values; or a list of discrete values (e.g., -5, 12, or 49).

I changed the LTspice color defaults to use the resistor color code, so the first trace is brown, then red, then orange, and so forth. When probing different parts of a circuit this makes it easier to keep track of which trace goes with which node in the circuit. With .step the same color sequence is also used, as shown in **Figure 2**.

Setting the waveform display colors to match the resistor code is easy: From the Tools menu select Control Panel, Waveforms (tab), then click Color Schemes. This brings up a dialog that lets you assign the color for each oscilloscope trace shown in the order you display them. Or in this case, the order that .step displays.

Gate Control Voltage	VCA Gain
0	60dB
-1	58dB
-2	56dB
-3	53dB
-4	48dB
-5	35dB
-5.227	20dB
-5.259	10dB
-6	0dB

Table 1: Most of the gain change for VCA #1 occurs near the negative voltage limit applied to the FET's gate.

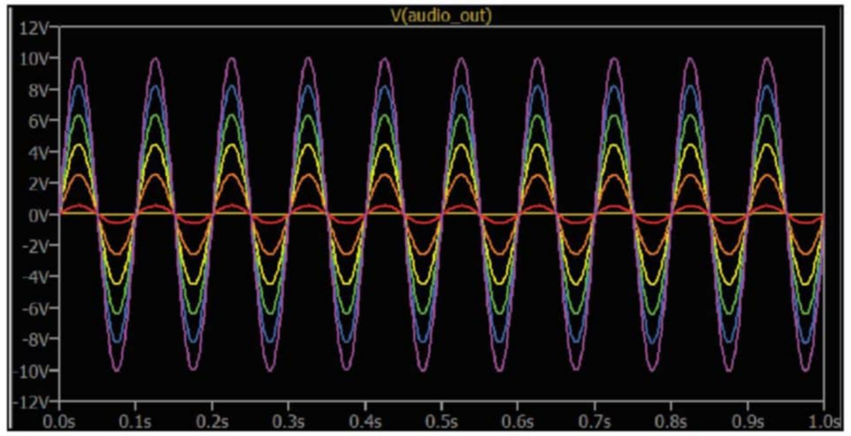


Figure 2: This shows the output of op-amp U1 when the .step command sequences the FET's gate voltage from -6V up to 0 volts in 1V increments.

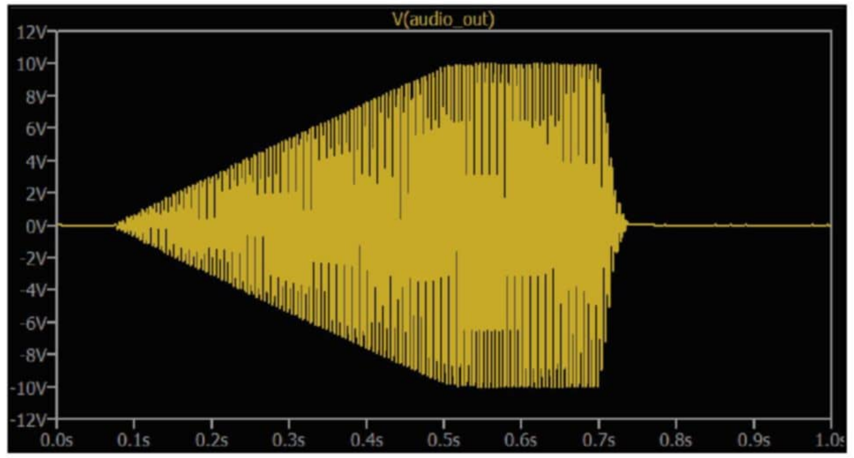


Figure 3: Connecting CV source V5 to the FET's gate instead of V4 displays a swept gain change over time, rather than the stepped increments shown in Figure 2.

Unfortunately, .step doesn't work in descending order, instead reversing the values to always be ascending. When I reported this problem in the LTspice Users Group, one of the experts there, Dave Bell, showed a simple work-around. In a circuit unrelated to this article I wanted to step resistor {X} backward from 21kΩ down to 1kΩ in 5kΩ increments. Dave demonstrated that changing {X} to {22k-X} forces the correct descending order. Note 22kΩ for the formula's resistor value versus 21kΩ for the starting .step value, because the last step is 1kΩ (22 to 21).

If you disconnect CV source V4 and connect V5, LTspice will display the gain change over time as shown in **Figure 3**. I set V5 to output a Pulse rather than a steady or stepped DC voltage. When a voltage source is set for Pulse you can control the initial voltage, rise time, destination voltage, number of cycles, and more. If you swap the CV sources as described, also change the audio sine wave to 200 cycles of 200Hz, change 0.1 in the .tran command to 1, then convert the .step command to a comment: Right-click the .step command line, click cancel, then you can select Comment to disable it. If you leave V4 set to {X} that will pop up a warning message you can ignore. Or change {X} to a number such as 3.

VCA #2

The core VCA design shown in **Figure 4** is a linear current multiplier. It multiplies the input current into the bases of Q1 and Q2 times the collector current set by R12. As the inverted control voltage pulls down more and more on their emitters, the transistors conduct an increasing amount. We usually see an NPN emitter connected to ground either through a resistor or directly, with the input signal pulling up on the base. In this case, the bases are connected to ground, and the emitters are pulled down toward a negative voltage.

Since this is a differential pair, the input feeds only one side because the signal is the voltage difference between the transistor bases. But to preserve the critical circuit balance it's important to use a dual transistor for Q1 and Q2, where both transistors are on the same die. There's no need for another figure showing this VCA's output because it's nearly identical to Figure 2.

Of course, a standard common-emitter single transistor amplifier stage also multiplies current, but this circuit's balanced design has several advantages. First, unlike the FET VCA, the gain change is perfectly linear, exactly tracking the amount of change in the control voltage. More important, as the control voltage varies the gain, the DC voltage on the transistor collectors shifts up and down in sync. By balancing the circuit with two equal collector currents feeding a differential op-amp input, the DC shift cancels out. So again the voltage difference at the collectors is the audio

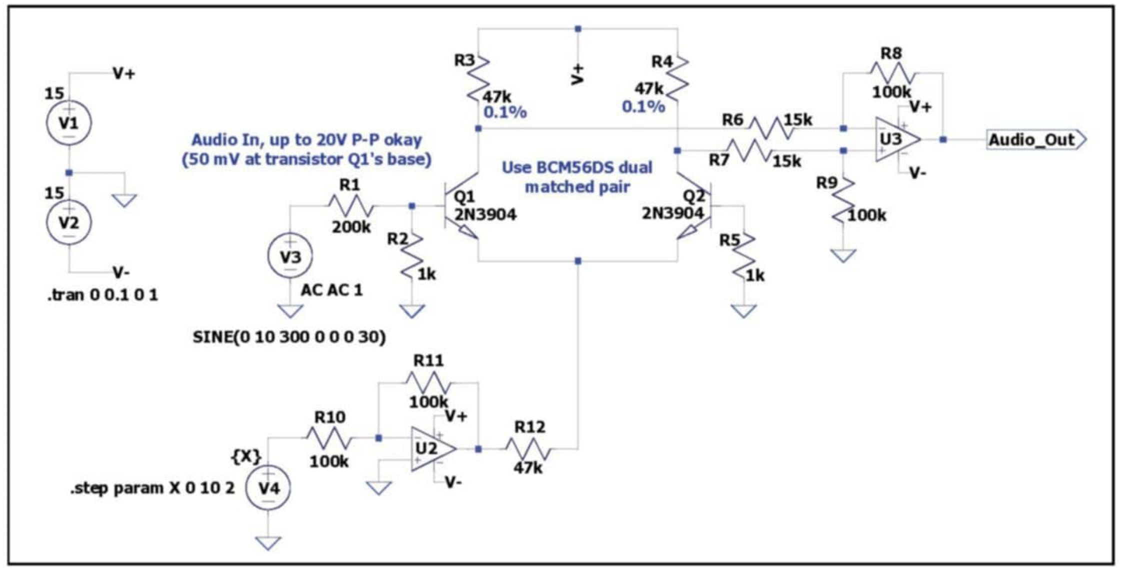


Figure 4: The balanced configuration of this multiplier circuit blocks the DC voltage shift that occurs with gain changes from getting to the audio output.

signal. Without this balancing act you'd get a thump when the control voltage changes, or a loud click if it changes quickly with a fast attack or decay from the ADSR.

This circuit is similar to the VCA that Robert Moog designed for his synthesizers, but was improved by Johan Verhoeven at Skull & Circuits to use a differential op-amp instead of a more complex arrangement of discrete transistors. I simplified the circuit further by removing some unnecessary components, and specifying 0.1% resistors for R3 and R4 to allow removing a trim pot. Readers interested in synthesizer circuits will enjoy the Skull & Circuits website. It's a great source of schematics, and also fully assembled modules. The link is listed in Resources.

VCF

The circuit shown in **Figure 5** is called a ladder filter because of the vertical arrangement of filter sections. **Figure 6** shows how the filter's response changes with different control voltages. The core circuit was designed by Robert Moog for his MiniMoog synthesizer, so I claim no credit for its elegant design! My only change was to alter the control voltage summing at the lower left to accommodate the various modules in my own guitar-controlled synthesizer. The patent for this filter expired decades ago, so it's now public domain for anyone to use for any purpose.

As with VCA #2, this ladder filter pulls down on the bottom set of transistor emitters to change the impedance of the transistors above. Each upper transistor pair serves as one filter pole, so all four stages set the filter roll-off to 24dB per octave. Positive feedback through potentiometer R7 boosts the output of the filter at its cut-off frequency, which also increases the Q of that boost. On a real MiniMoog, the resonance can be increased enough to cause the filter to oscillate. The maximum amount of feedback available is set by R8, so you might need to reduce (or increase) its value slightly to resonate only at the very highest setting of R7. Note that the original Moog schematic specs R7 with a reverse audio taper, but a linear taper pot will be much easier to find.

This circuit is also balanced to avoid thumps and clicks at the audio output when the control voltage changes the frequency. As with VCA #2, matched transistor pairs are used to preserve the critical balance, but they're needed only for the bottom pair below the ladder. Again, the BCM56DS is an excellent and affordable choice. It probably wouldn't take much effort to replace the discrete transistor-based differential amplifier with an op-amp to save a few dozen components, as Johan Verhoeven did for VCA #2. But I left the schematic as is for its historical value, and to avoid inadvertently changing the filter's highly esteemed sonic character.

The last item worth explaining in this filter circuit is Keyboard Tracking, whose voltage input is at the bottom left of Figure 5. Normally, a filter is swept automatically by its own ADSR as each note is played, or left at a static position, or varied manually by the musician for expression. Most analog synthesizers also have an option for the filter to track the keyboard, so playing higher notes shifts the filter frequency higher in addition to any sweeping or LFO effects. For example, if you have a resonant filter boost that brings out the third harmonic of a note, you'll want the filter to track those notes as you play up and down the keyboard. Keyboard tracking also makes possible sound effects such as wind noise that can be varied in pitch or even played as a melody.

In order for the filter to precisely track the pitch of the current note, a potentiometer is needed. This could be an internal trim-pot on the filter's circuit board, or a front panel control that lets the user scale the amount of tracking. But generally, when a note rises or lowers one octave, the filter's cut-off frequency should go up or down the exact same amount. As noted on the schematic, R22 is

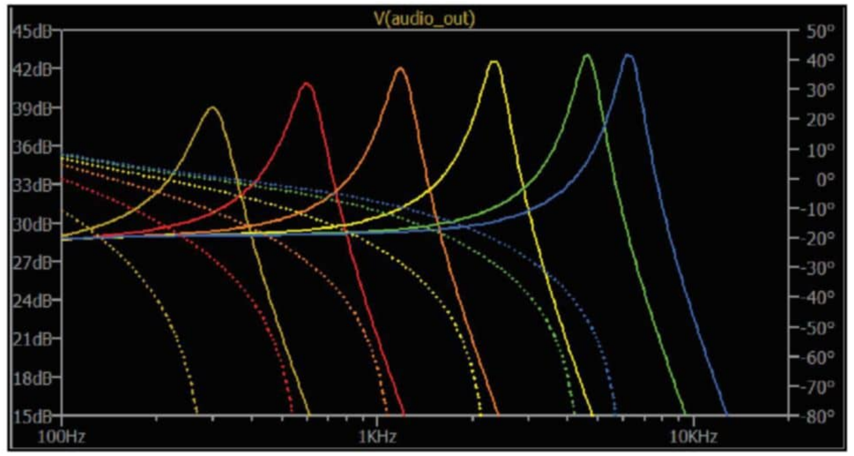


Figure 6: This graph shows the stepped response with Resonance fairly high by setting resistor R7 to 500Ω. The dotted lines show the phase at each step, and can be ignored since what we hear is only the response.

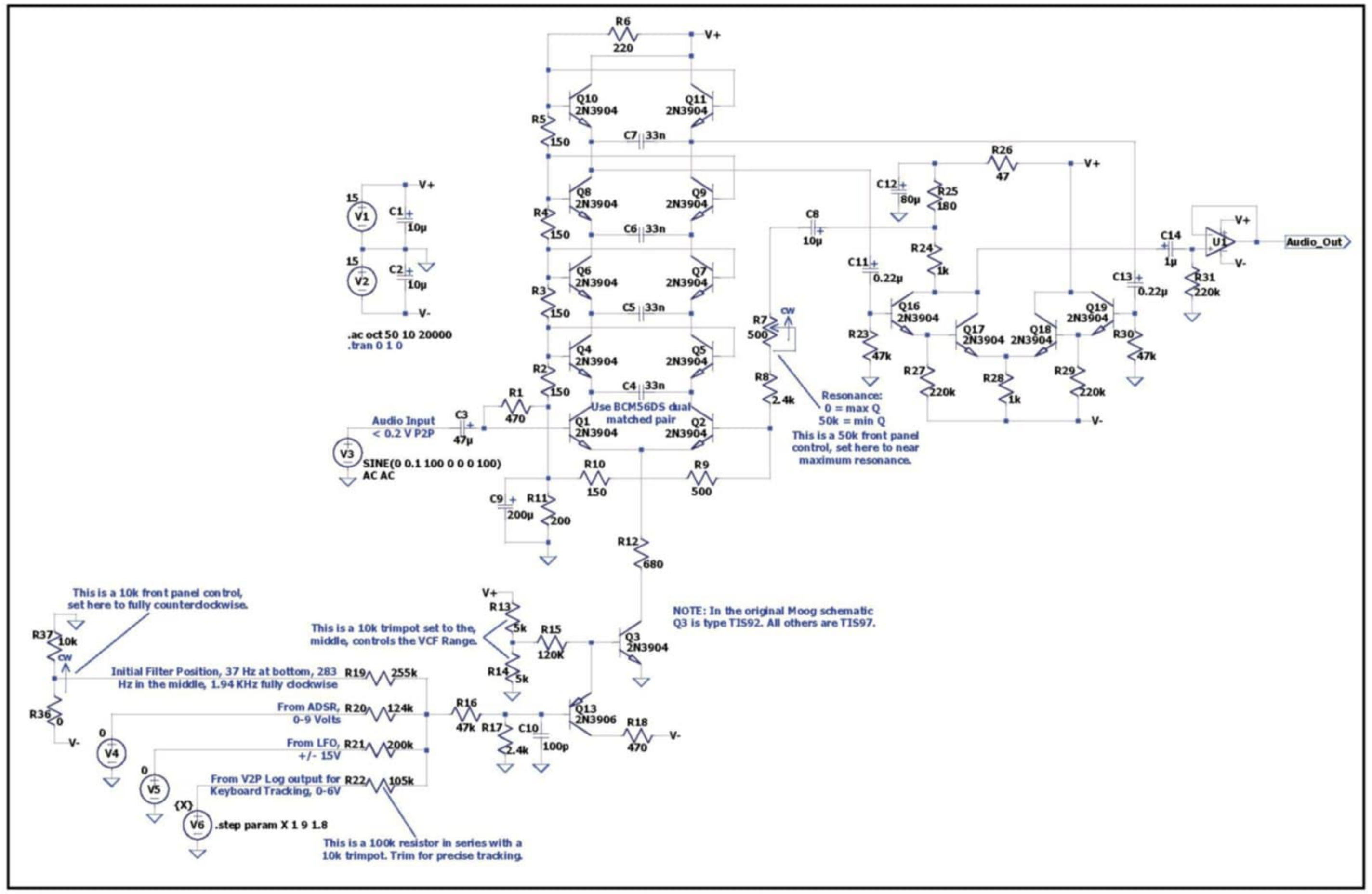
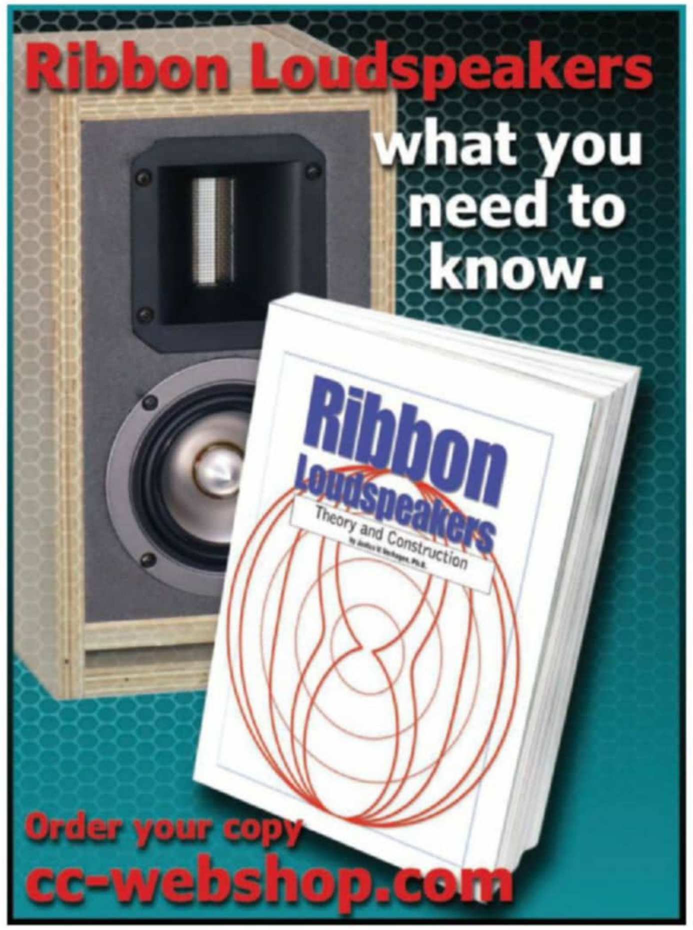


Figure 5: This is the original 24dB/octave resonant low-pass filter designed by Robert Moog in the 1960s.



actually a fixed resistor in series with a trimmer to precisely set a 1-to-1 ratio of note pitch versus filter frequency.

Finally, if you'd like to dig into the circuit's internal operation you can switch the display from a stepped frequency response to the oscilloscope view. First, right-click the command line that currently specifies AC analysis:

```
.ac oct 50 10 20000
```

Then Cancel, then select Comment to disable it. Next, right-click this command which is now a comment, to enable Transient Analysis:

```
.tran 0 1 0
```

Then select "SPICE directive" to enable it. As with VCA #2, you should also disable the .step command, and change {X} to a number between 0 and 10 to set the filter's cut-off to a single frequency manually.

About the Author

Ethan Winer has been an audio engineer and professional musician for more than 45 years. His Cello Rondo music video has received nearly 2 million views on YouTube and other websites, and his book *The Audio Expert* published by Focal Press, now in its second edition, is available at amazon.com and Ethan's own website ethanwiner.com. Ethan is also a principal at RealTraps, a US manufacturer of high-quality acoustic treatment.

Project Files

To download the links for LTspice VCA circuit file: [VCA-FET.asc]; the LTspice VCA circuit file: [VCA.asc]; and the LTspice VCF circuit file: [VCF.asc], visit: http://audioxpress.com/page/audioXpress-Supplementary-Material.html

Recources

LTspice, www.analog.com/en/design-center/design-tools-and-calculators/ltspice-simulator.html

LTspice online support forum, https://groups.io/g/LTspice

National Semiconductor's Applications Notes handbooks, https://archive.org/details/manuals-nationalsemiconductor

Skull & Circuits, www.skullandcircuits.com

E. Winer, "Building a Guitar-Controlled Synthesizer: The Sample & Hold Time Machine," *audioXpress*, April 2022.

E. Winer, "Building a Guitar-Controlled Synthesizer: Frequency to Volts and Back," *audioXpress*, May 2022.

E. Winer, "Building a Guitar-Controlled Synthesizer: LFO and ADSR," *audioXpress*, June 2022.

E. Winer, "Building a Guitar-Controlled Synthesizer: Input Section & Pick Detector," *audioXpress*, July 2022.

Al Fine

Even if you don't intend to build an entire analog synthesizer, the circuits shown throughout this series can serve as "cookbook" elements to use in other projects. When I started learning electronics in the 1960s there was no Internet. By the 1970s, I was in my 20s working as a technician at electronic companies, so I had access to real engineers who were kind enough to answer my beginner questions. I also got to read every issue of *Electronics* and *EDN*, excellent magazines offered for free to electronics engineers. In every issue both magazines offered clever circuit designs and interesting hacks, as well as industry news and product reviews. The National Semiconductor books mentioned earlier are another fabulous resource for "cookbook" style building blocks. Today, you can Google the name of a circuit type and come up with myriad circuits ranging from elegant to incompetent.

My point is that circuit elements like those presented in this series can be used and reused again and again. The Period-to-Voltage-to-Period article showed two types of Log converter circuits, a Sample & Hold, and simple ways to create octaves using basic digital logic chips. The LFO/ADSR article explained hysteresis and showed how to add that to a voltage comparator. The Input Section article shows how to make a noise gate, as well as a "no-drop" full-wave rectifier that avoids losing 0.6V across the diodes. That article also included a clever "cheat" showing how to easily select capacitor values for the 555 timer IC. This article shows how to make two types of voltage controlled amplifiers plus the classic Moog low-pass filter.

This is how I learned how to design electronic circuits, by piecing together smaller building blocks from magazines, and the application notes from semiconductor makers. I don't have a degree in electronics, and I'm no math expert! But I can assemble circuit elements and eventually come up with something that works. Online calculators for every type of filter are readily available, saving you time and effort to find suitable resistor and capacitor values. In the old days you'd have to solve formulas such as:

$$f_c = 1 / (2 \times \pi \times R \times C)$$

again and again as you experiment with different frequencies in a high-pass filter. The LTspice software I used in all of these articles is vastly easier and faster—and much less expensive!—than actually building and modifying circuits and owning a bench full of test gear. Indeed, life today is good for electronics hobbyists.

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